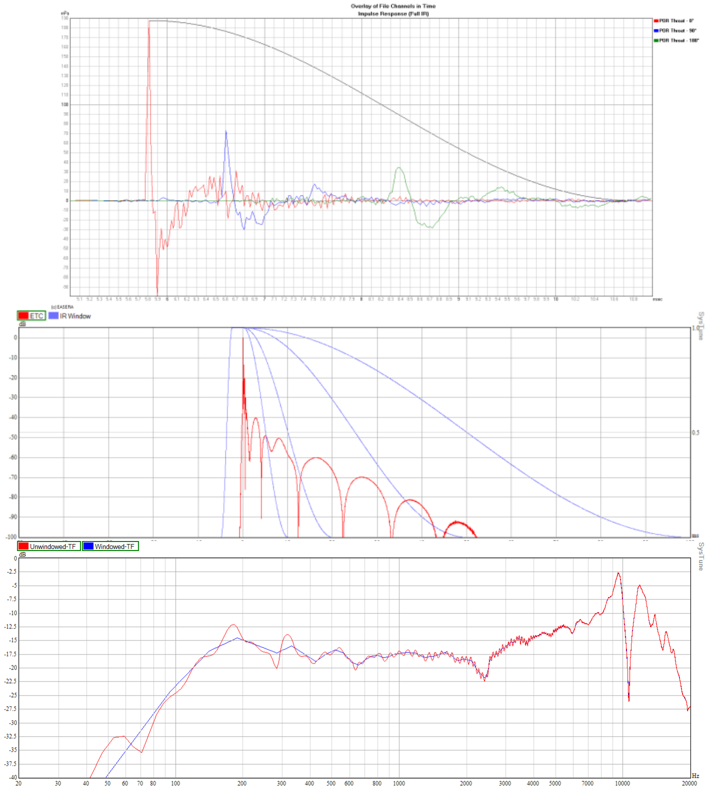
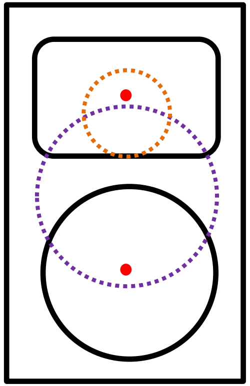
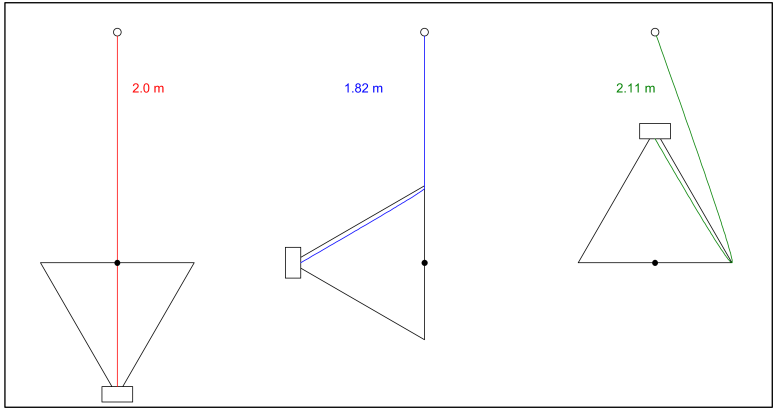
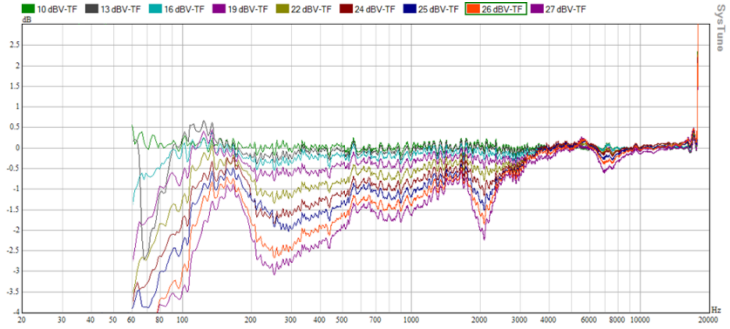
Measuring Loudspeaker Systems









Topics

1) FFT Basics
2) Windowing (Signal Acquisition; Impulse Response & Frequency Resolution)
3) Signal to Noise Ratio
4) Measurement Distance
5) Directivity Measurements (Point of Rotation; Angular Resolution)
6) Ground Plane Measurements
7) Impedance
8) Maximum Input Voltage
9) IR Arrival Time & Synchronization of Pass Bands

Measuring Loudspeaker Systems

Please stop me at any time for questions.

A PDF of all the slides will be available next week at my website
www.excelsior-audio.com

FFT Measurement Basics

FFT – Fast Fourier Transform

Mathematical operation that allows ***time domain*** data (a recorded signal) to be transformed to the ***frequency domain*** (spectral content of the signal)

FFT Measurement Basics

FFT Data Block

To perform an FFT we need a block of data for the analysis. This FFT block has a certain size associated with it. It is typically in binary increments (2^n) to make computer-based calculations faster.

$$FFT\ Block\ Size\ =\ Number\ of\ Samples$$

(Typical values 2,048; 4,096; 8,192; 16,384; 32,768; etc.)

FFT Measurement Basics

Sample Rate

The sample rate we chose determines the high frequency limit for our measurement.

$$HF\ Limit \leq \frac{Sample\ Frequency}{2}$$

(Typical values 44.1 kHz, 48 kHz, 88.2 kHz, 96 kHz, etc.)

FFT Measurement Basics

Sample Rate

The sample rate can also be expressed as a period.
This is the reciprocal of sample rate.

$$Sample\ Period = \frac{1}{Sample\ Frequency}$$

(Typical values 22.7 us, 20.8 us, 11.3 us, 10.4 us, etc.)

FFT Measurement Basics

Frequency Resolution

The FFT block size and the sample rate determine the frequency resolution and the LF limit of the measurement.

$$FFT\ Size\ =\ Sample\ Period\ *\ Number\ of\ Samples$$

(FFT Size = 20.8 us * 8,192 samples = 0.17 s)

$$\Delta frequency\ =\ 1/\Delta time$$

Freq. Res. = 1 / 0.17 s = 5.9 Hz

FFT Measurement Basics

FFT Bins

In the frequency domain the FFT "fills" each discrete bin with the energy contained in the time domain signal.

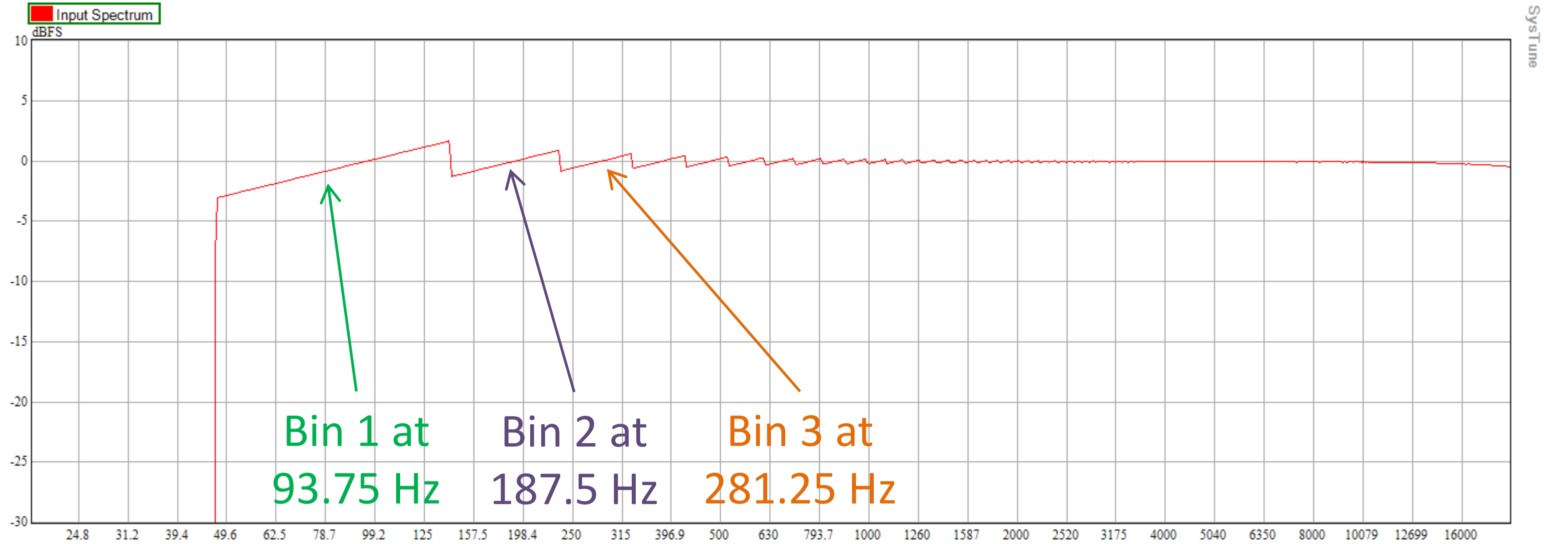
Each FFT bin is the same width as the frequency resolution. They are spaced linearly.

The energy can only go into a bin. It can't go to frequencies between the bins.

FFT Measurement Basics

FFT Bins – 93.75 Hz wide spaced every 93.75 Hz

48 kHz Sample Rate	512 Sample FFT Size	93.75 Hz Resolution



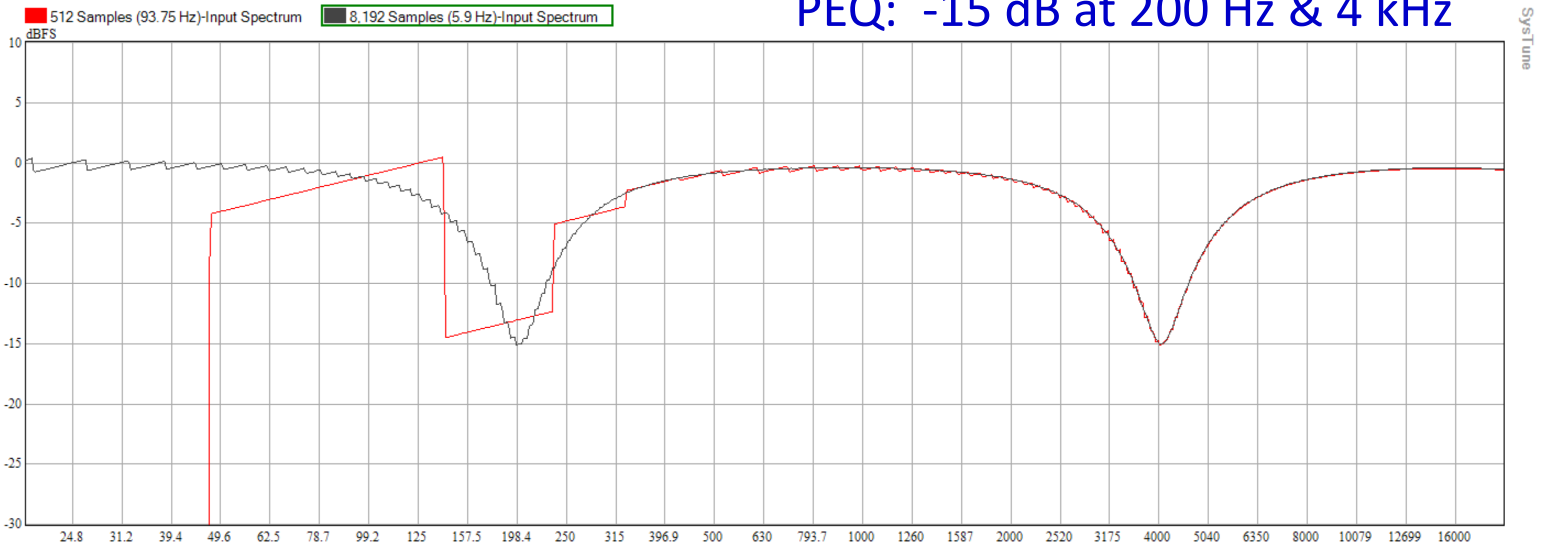
FFT Measurement Basics

Single Channel FFT

One signal is analyzed and its spectrum (frequency domain) is displayed.

The excitation signal is part of what is seen in the measurement results.

PEQ: -15 dB at 200 Hz & 4 kHz



FFT Measurement Basics

Dual Channel FFT

Two signals are analyzed:

Measurement channel and Reference channel

$$FFT\left[Deconvolution\left(\frac{Measurement}{Reference}\right)\right]\xrightarrow{\text{yields}} Transfer\ Function$$

(or Frequency Response)

Deconvolution in the time domain is the same as division in the frequency domain.

Because of the division (reference channel), the spectral content of the excitation signal is not seen in the measurement results.

FFT Measurement Basics

Transfer Function (TF)

The transfer function is complex, meaning it contains both magnitude and phase.

The TF (frequency domain) and the Impulse Response, or IR, (time domain) are different views of exactly the same thing.

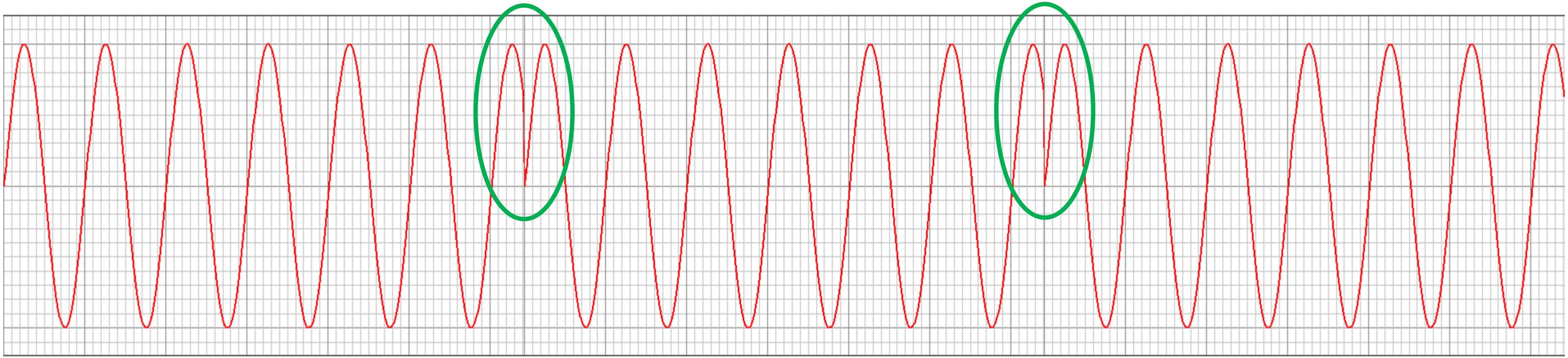
The frequency domain allows us to see ***what*** is happening.
The time domain allows us to see ***when*** it is happening.

Uncertainty Principle: $\Delta frequency = 1/\Delta time$

FFT Measurement & Windowing

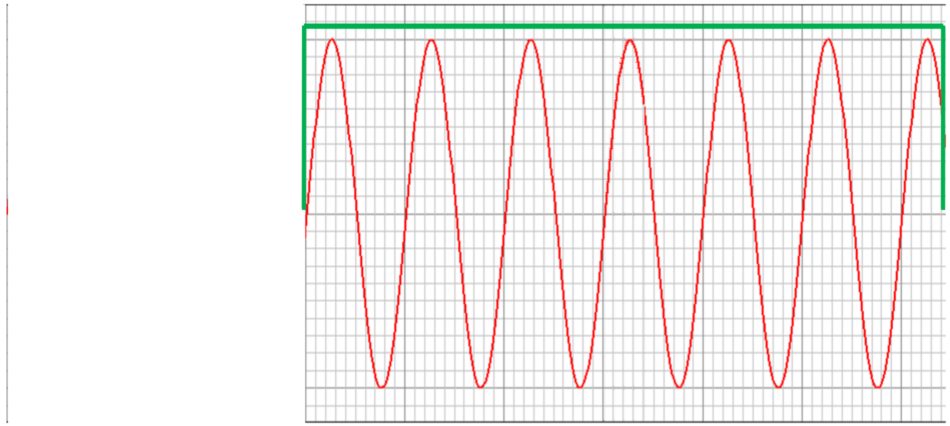
The Fourier Transform requires that the signal be infinite. For the FFT we assume the signal is infinite by "looping" it.

But there are consequences.



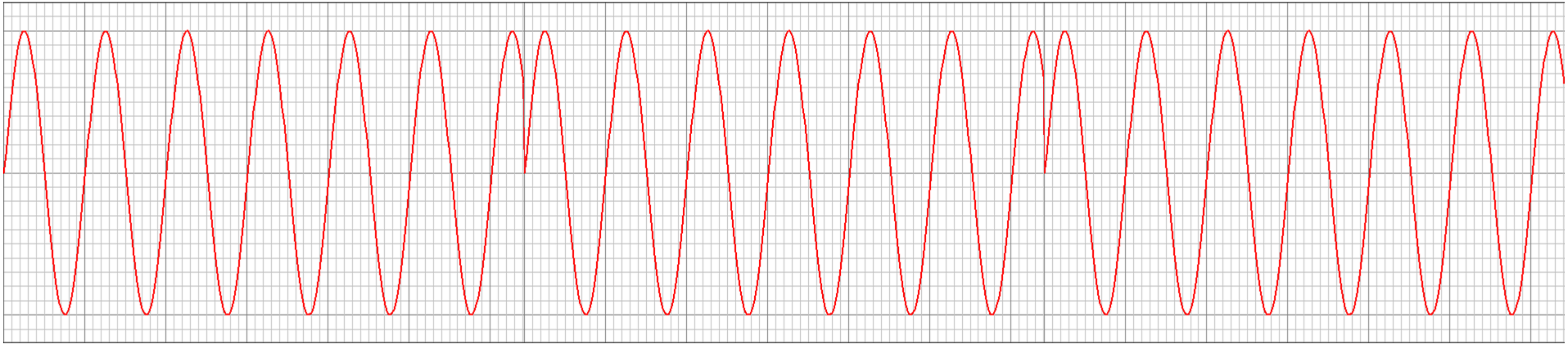
FFT Measurement & Windowing

Just the act of taking an FFT block is applying a rectangular window to the data. This is a ***Signal Acquisition*** window.
(not an ***Impulse Response*** window)



FFT Measurement & Windowing

This signal acquisition window can lead to our discontinuity problem.

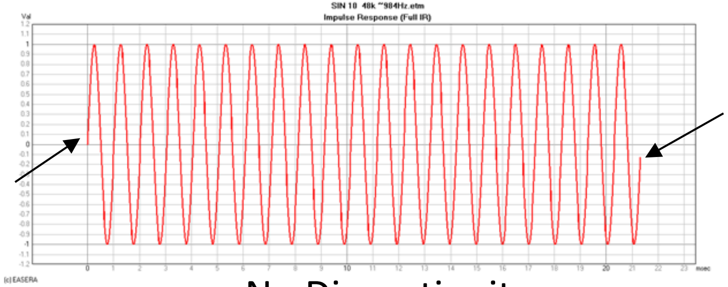


FFT Measurement & Windowing

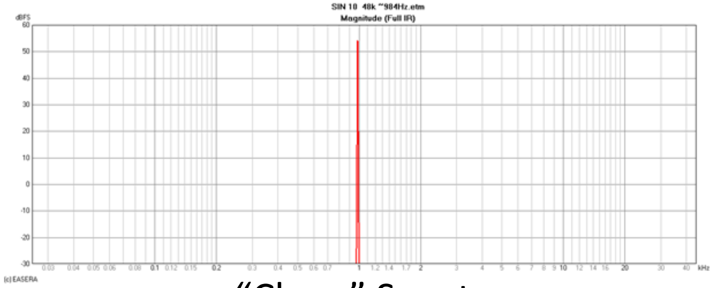
Example: 48kHz, 1024 Samples

997 Hz Sine

An integer number of periods ***does*** fit within the FFT block



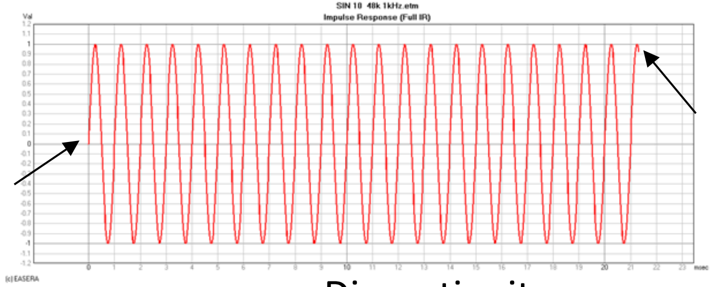
No Discontinuity



"Clean" Spectrum

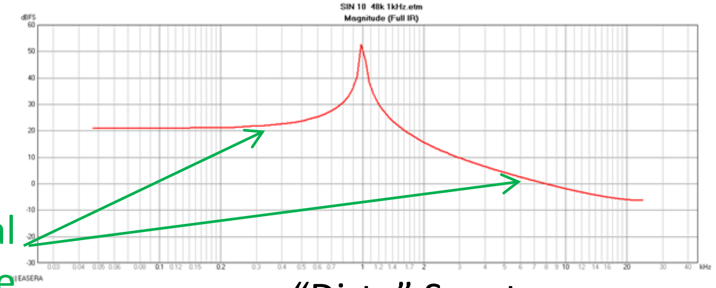
1,000 Hz Sine

An integer number of periods ***does not*** fit within the FFT block



Discontinuity

Spectral Leakage

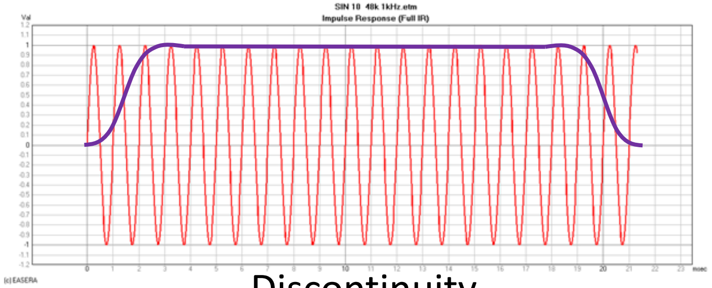


"Dirty" Spectrum

FFT Measurement & Windowing

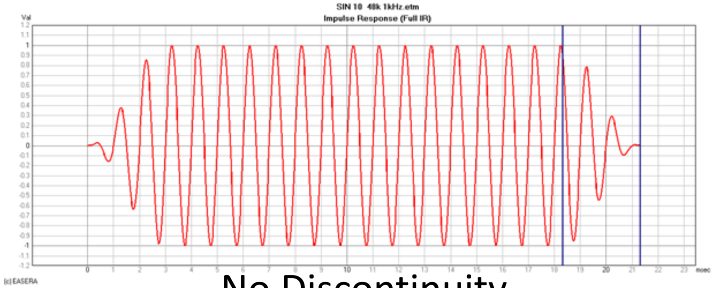
Example: 48kHz, 1024 Samples

1,000 Hz Sine



Discontinuity

Windowed 1,000 Hz Sine

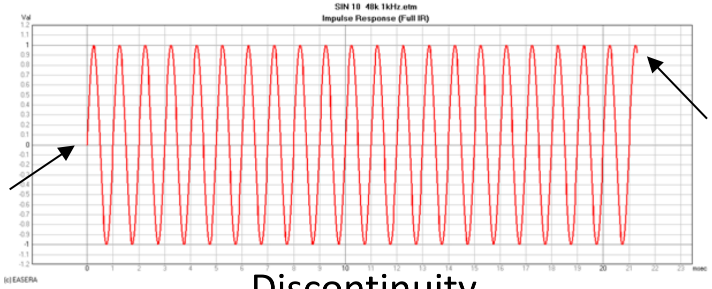


No Discontinuity

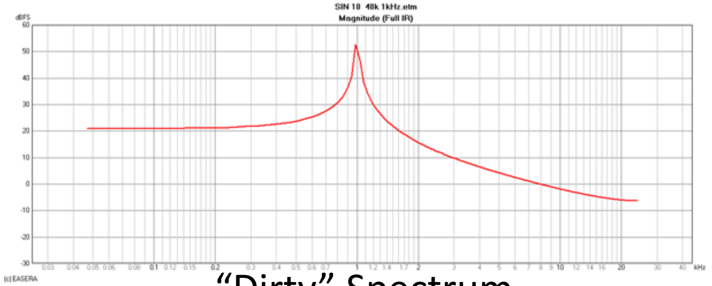
FFT Measurement & Windowing

Example: 48kHz, 1024 Samples

1,000 Hz Sine

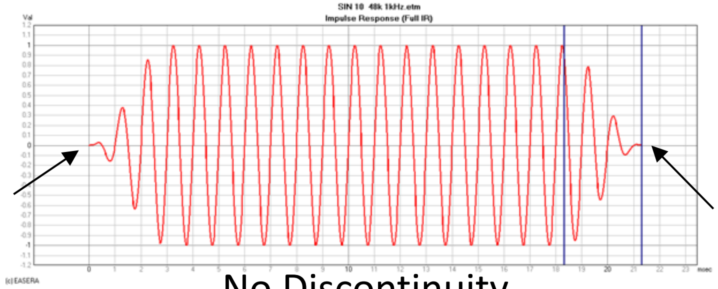


Discontinuity

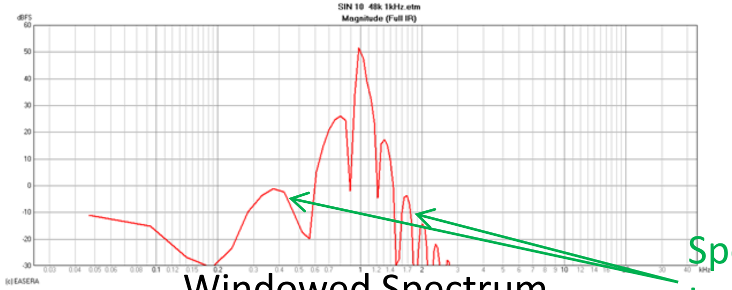


"Dirty" Spectrum

Windowed 1,000 Hz Sine



No Discontinuity



Windowed Spectrum

FFT Measurement & Windowing

Advantages of windowing

- Artifacts (spectral leakage) due to FFT block discontinuities can be reduced by using non-rectangular windows.

Disadvantages of windowing

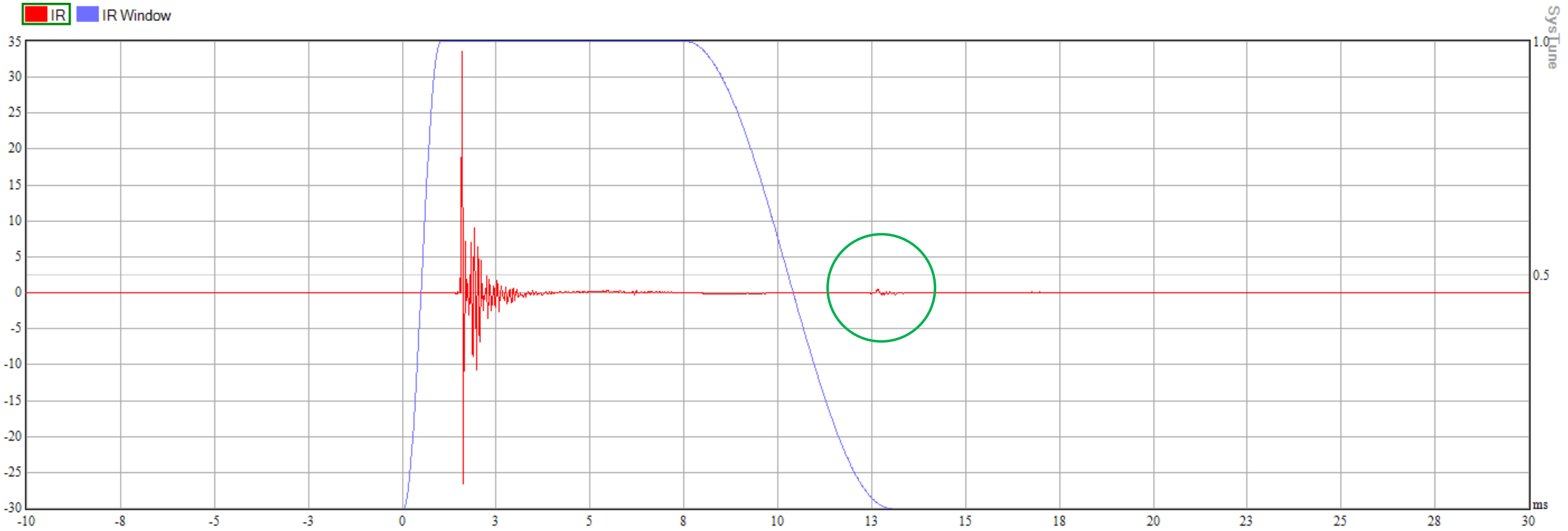
- Windowing can't completely eliminate spectral leakage (side lobes).
- Information is reduced or lost due to the window.

One Solution

- Using same FFT size for source signal and the analysis block can greatly reduce or eliminate these problems.

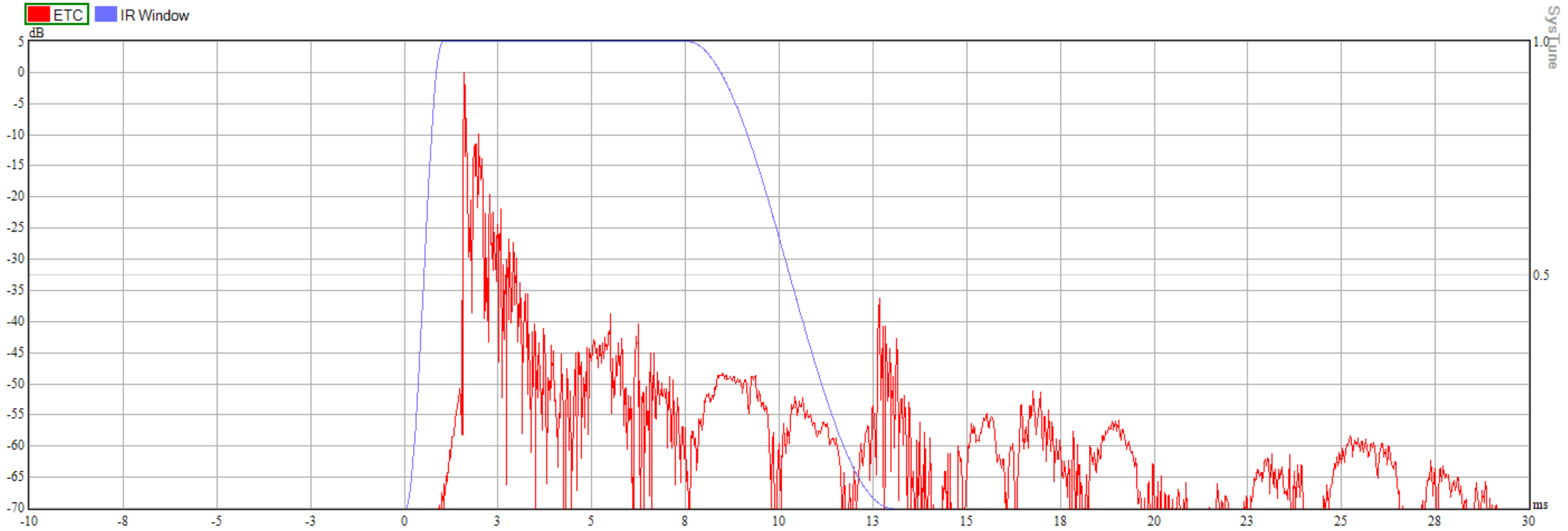
Impulse Response Windowing

An ***IR*** window is different than a ***Signal Acquisition*** window



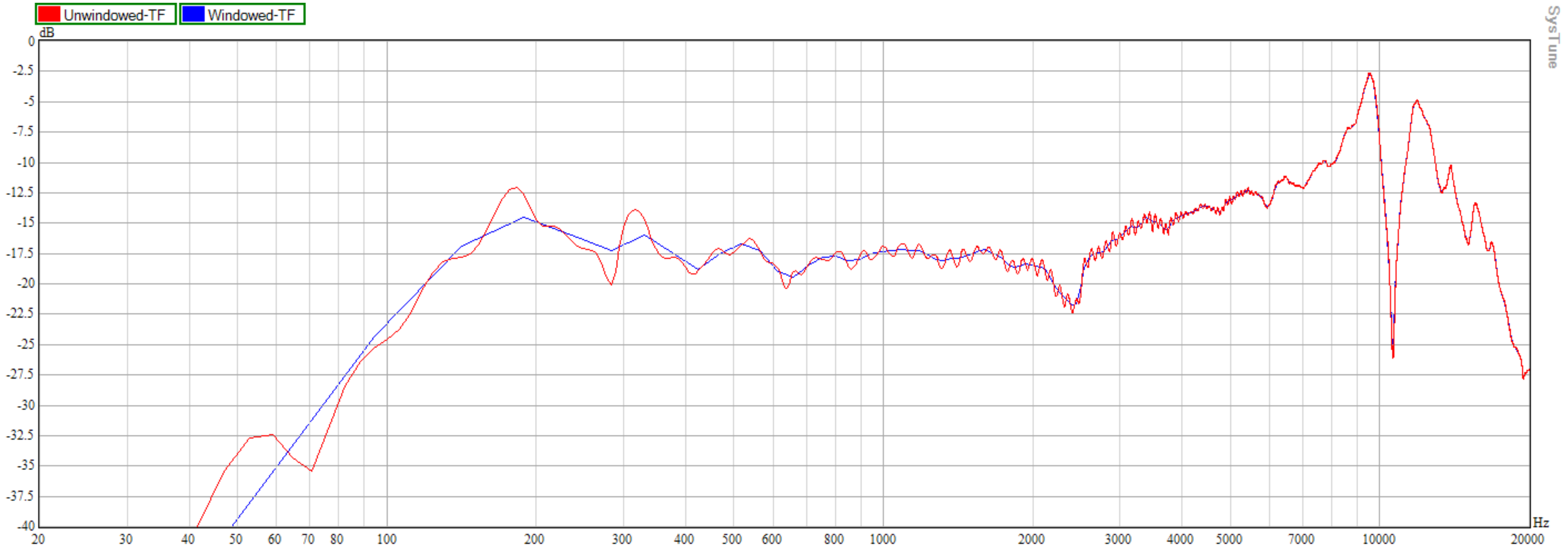
Impulse Response Windowing

An *IR* window is different than a *Signal Acquisition* window



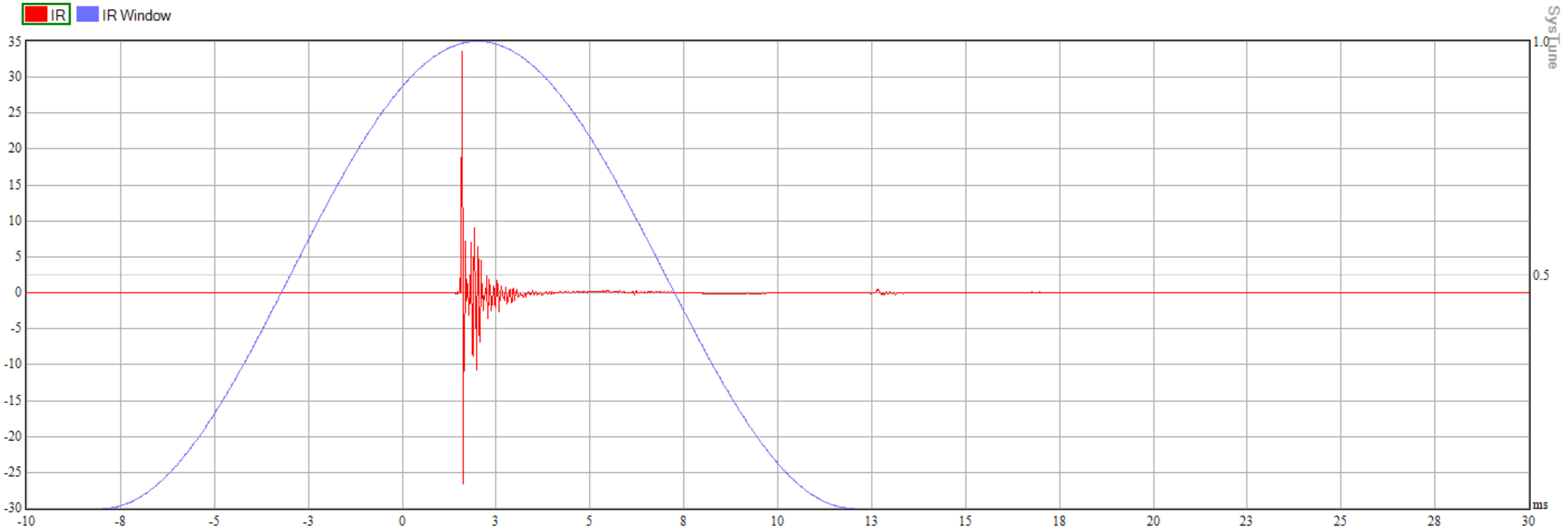
Impulse Response Windowing

Windowed and unwindowed transfer function



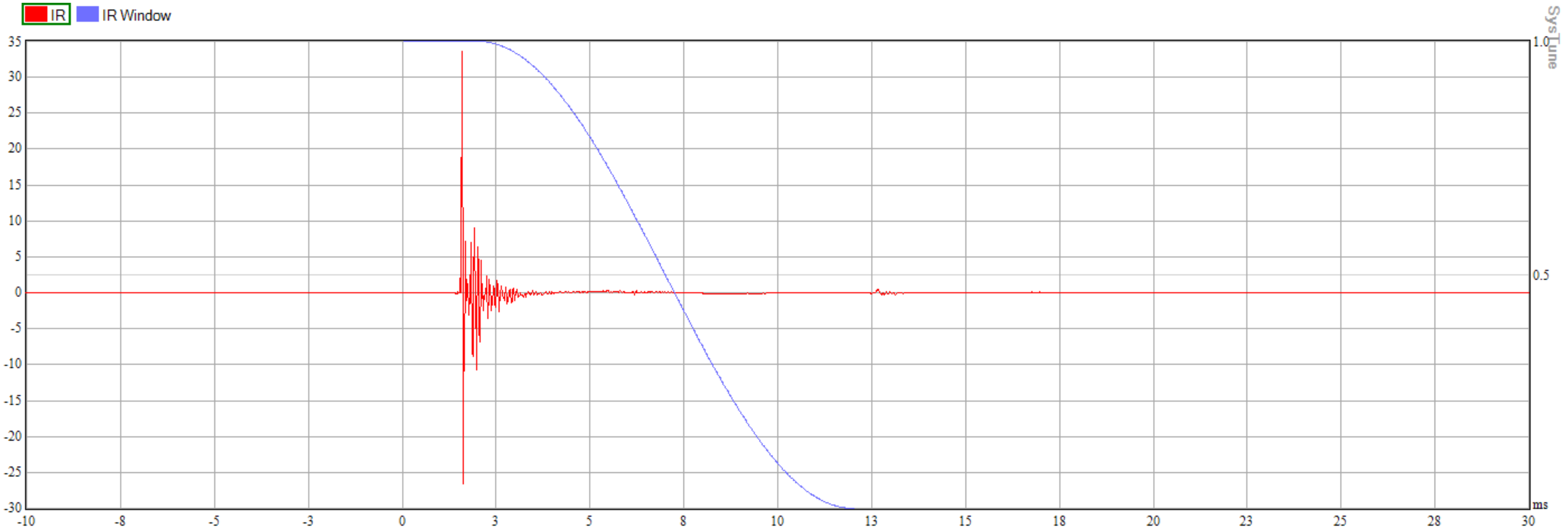
Impulse Response Windowing

Symmetrical Windows compared to Half-Windows



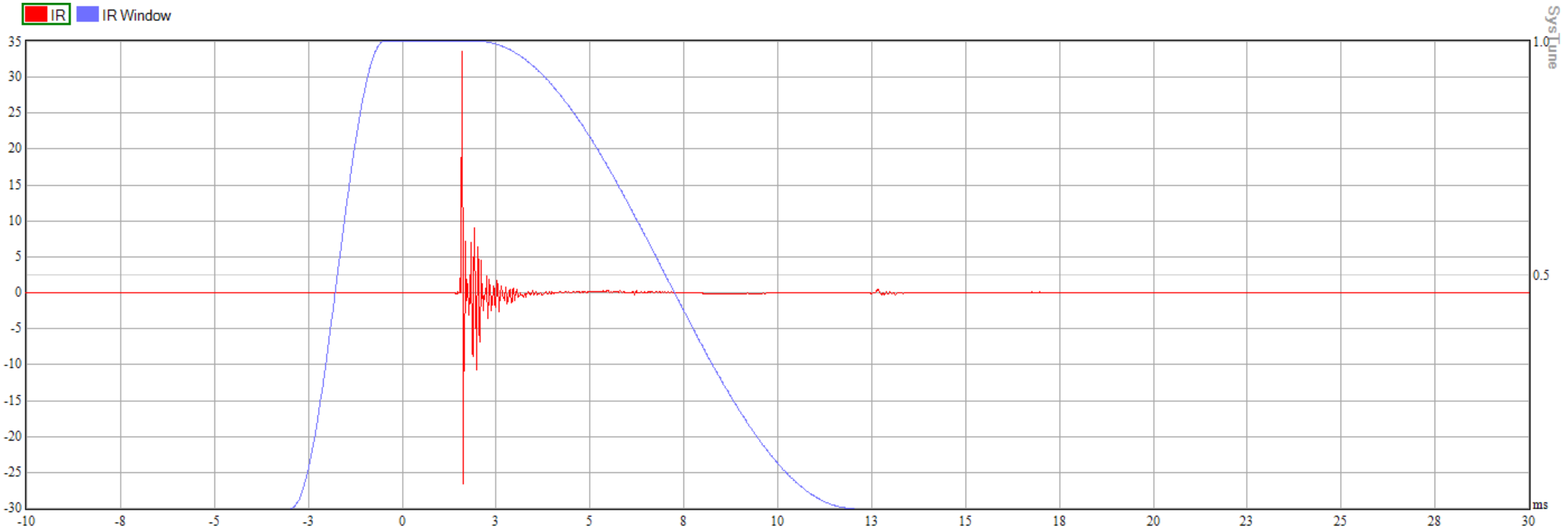
Impulse Response Windowing

Symmetrical Windows compared to Half-Windows



Impulse Response Windowing

Asymmetrical Left and Right Half-Windows

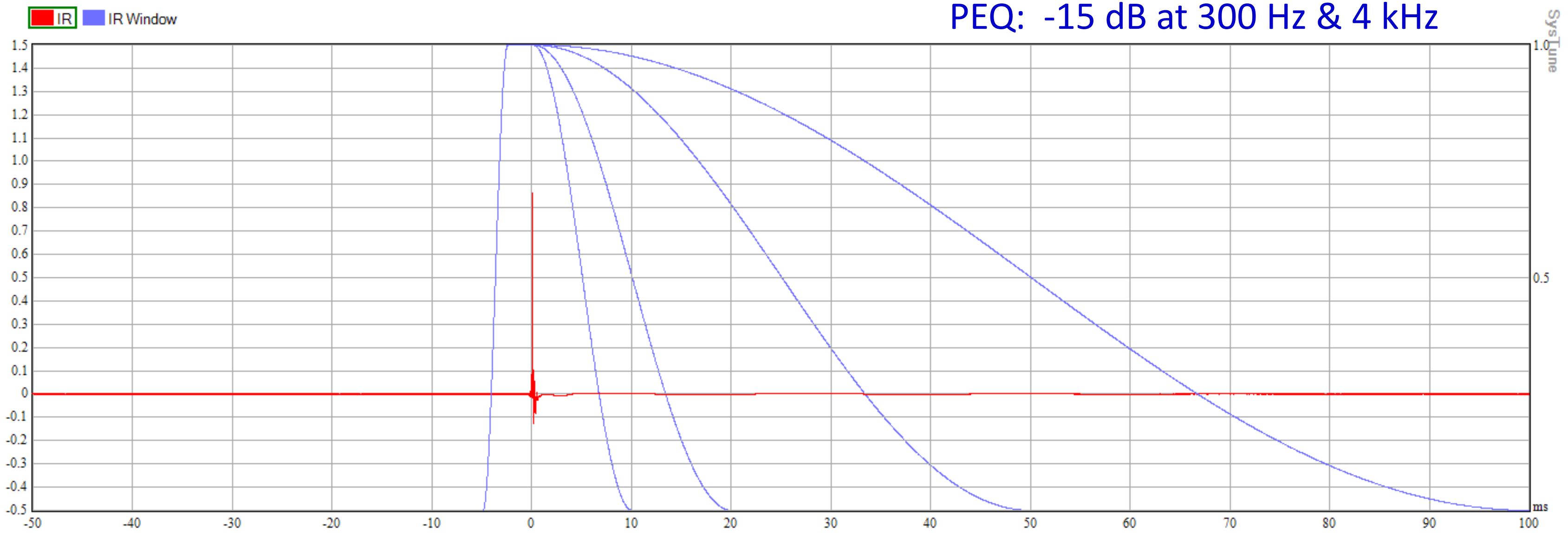


Impulse Response Windowing

Window length (time) affects frequency resolution

HP: 50 Hz, 24 dB/oct.

PEQ: -15 dB at 300 Hz & 4 kHz



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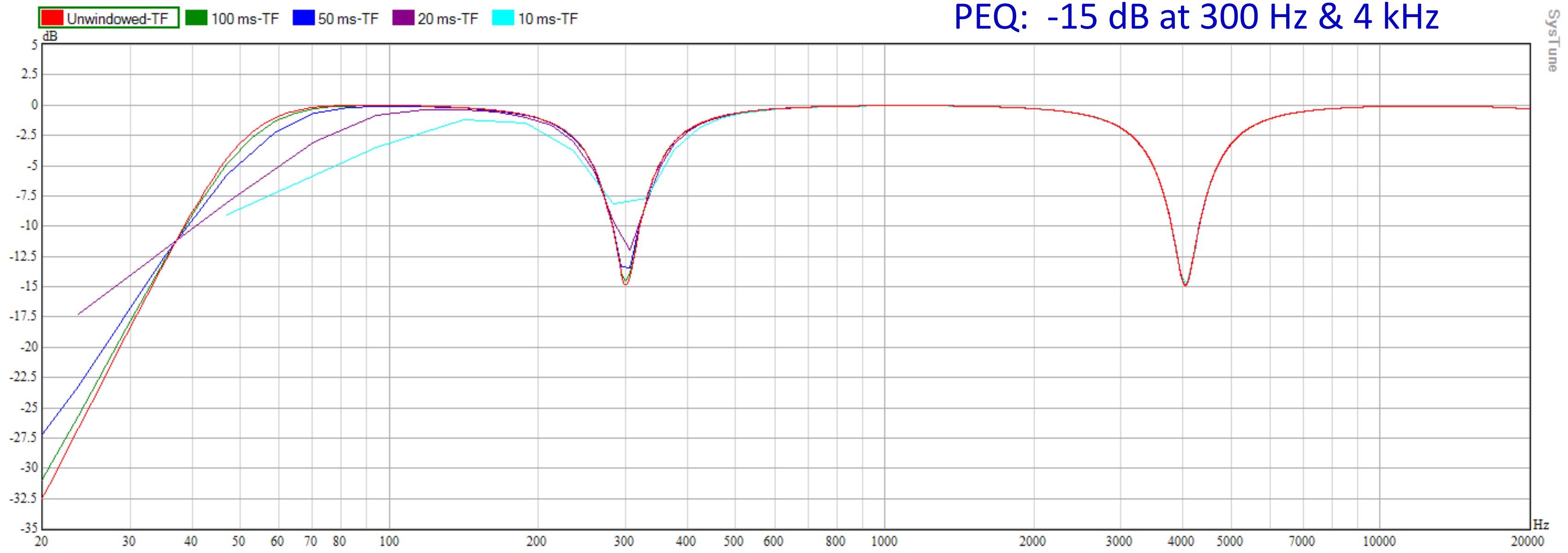

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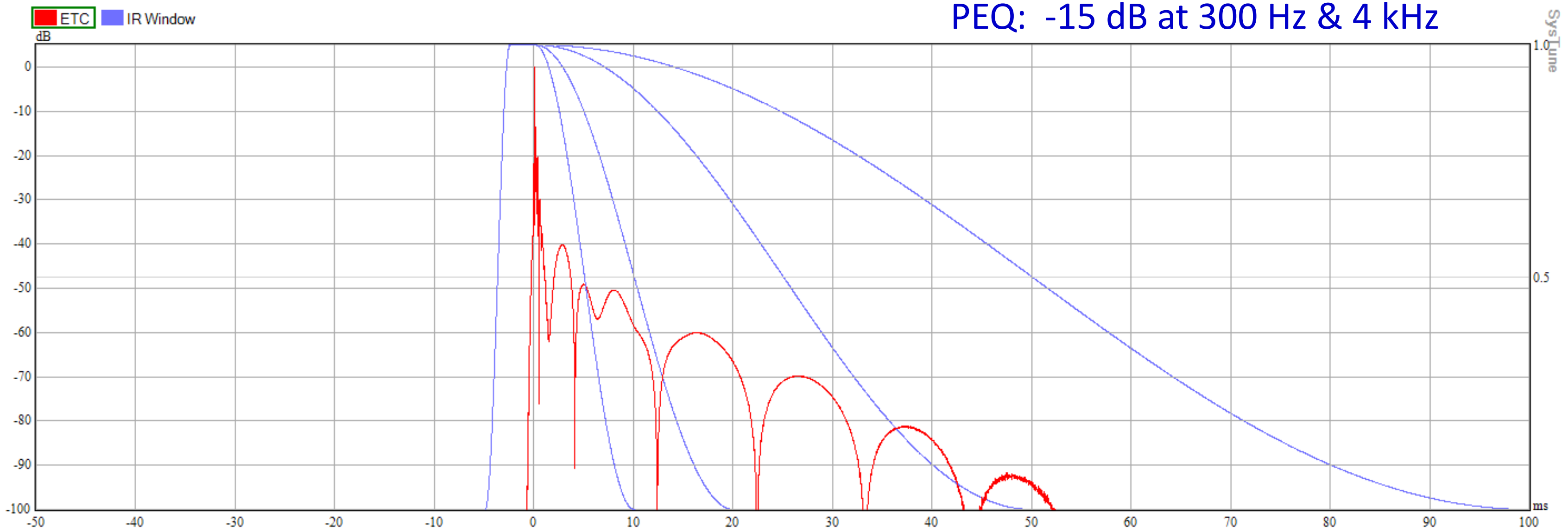


Signal to Noise Ratio

Everything that is not the desired signal is "noise"

- Ambient background noise
- Reflections *(only affected by IR windowing or TDS, not any of the other items below)*
- Electrical noise in test equipment (power amplifier, mic pre-amp, etc.)

Ways to improve S/N

- Brute Force – High SPL from DUT
- Time Delay Spectrometry (TDS)
- Log-Swept Sine
- Multiple Averages

For directivity measurements the SPL at off-axis positions may be greatly reduced from the on-axis SPL at certain frequencies.

Measurement Distance

How far away from the DUT should the measurement mic be placed? *Shouldn't this just be 1 meter?*

Measurements should be made in the far-field of the DUT.

High frequency limit for near-field to far-field transition

For line (or line-like) sources:

$$Distance \geq h^2 * {}^{f}/_{2c}$$

h is the height (length) of the line source
f is the frequency of interest
c is the speed of sound

Measurement Distance

How far away from the DUT should the measurement mic be placed? *Shouldn't this just be 1 meter?*

Measurements should be made in the far-field of the DUT.

High frequency limit for near-field to far-field transition

A modified version of this equation also holds well for non-line sources (pistons, cones, horns, etc.):

$$Distance \geq S * {}^{f}/_{c}$$

S is the surface area of the source

f is the frequency of interest

c is the speed of sound

Note that S should include the entire front of the loudspeaker (not just the drivers) to account for the diffraction at the edges of the enclosure.

Measurement Distance

How far away from the DUT should the measurement mic be placed? *Shouldn't this just be 1 meter?*

Measurements should be made in the far-field of the DUT.

Low frequency limit for near-field to far-field transition

$Distance \geq \lambda$ or $Distance \geq {}^{f}/_{c}$

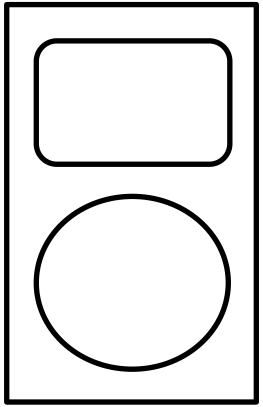
λ is the wavelength for the frequency of interest
f is the frequency of interest
c is the speed of sound

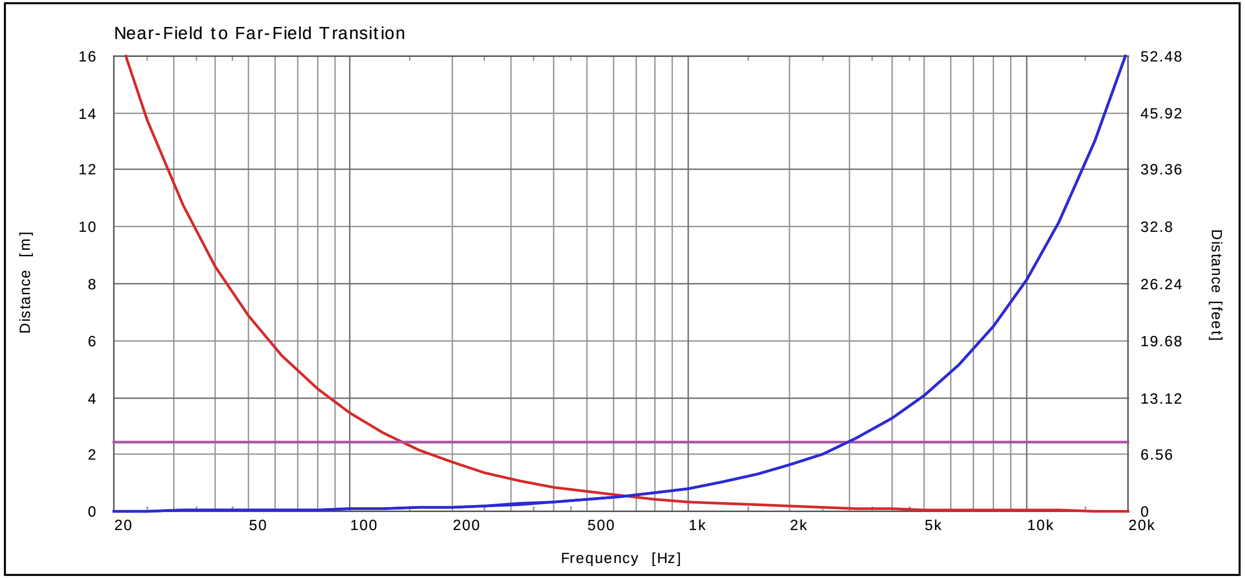
Measurement Distance

Example

$Distance \geq \lambda$

$Distance \geq S * {}^{f}/_{c}$





A rough approximation of 3x the largest dimension (diagonal) for the DUT is sometimes used.

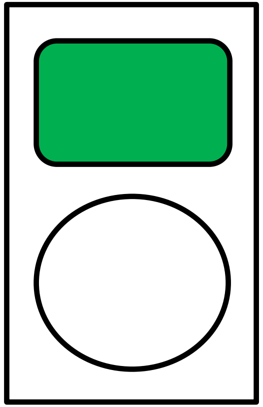
Diagonal is 0.81 m (2.64 ft.) 3x Diagonal is 2.4 m (7.9 ft.)

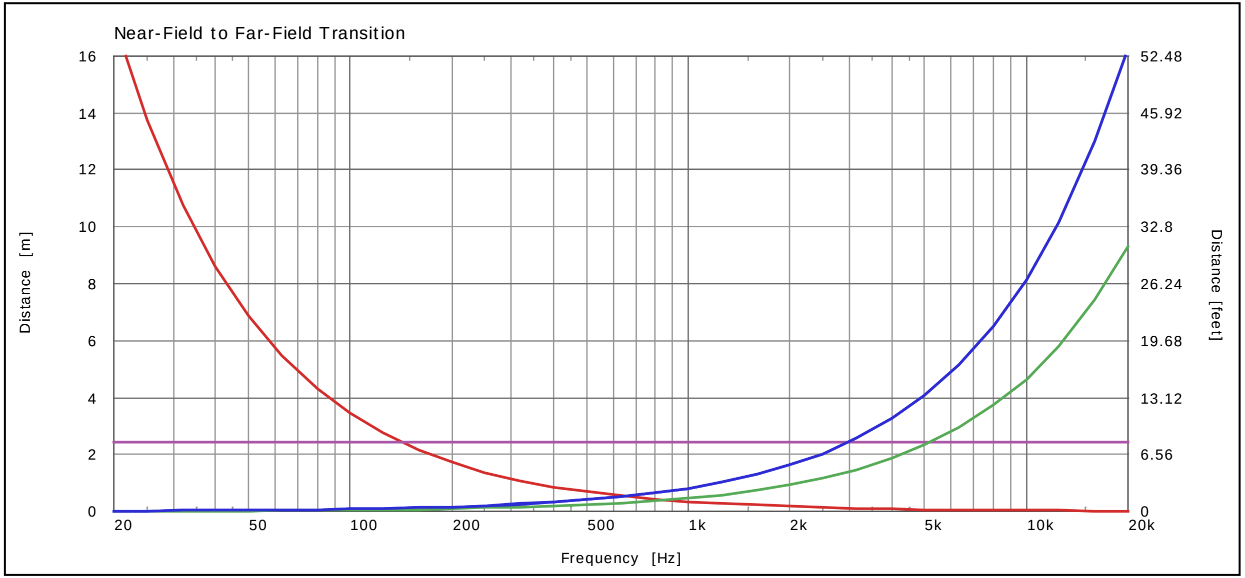
Measurement Distance

Example

$Distance \geq \lambda$

$Distance \geq S * {}^{f}/_{c}$





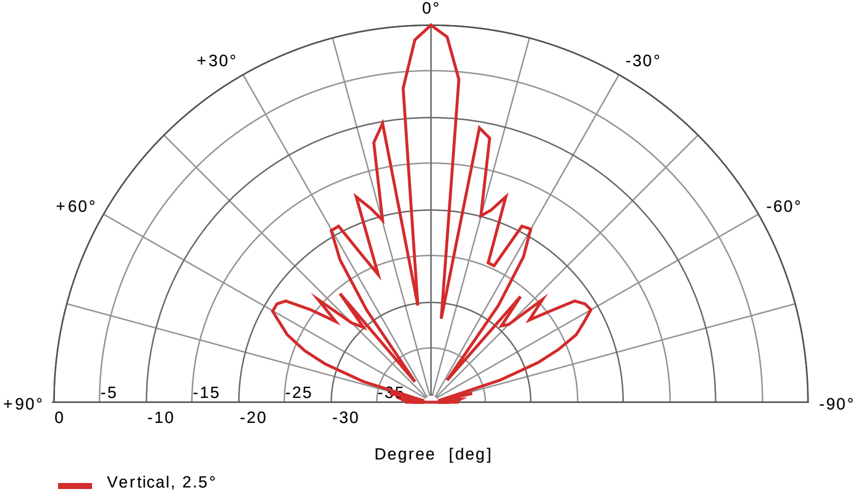
At higher frequencies where the HF horn has sufficient directivity control the size of the source, S, is only the horn. The bottom part of the cabinet can be neglected.

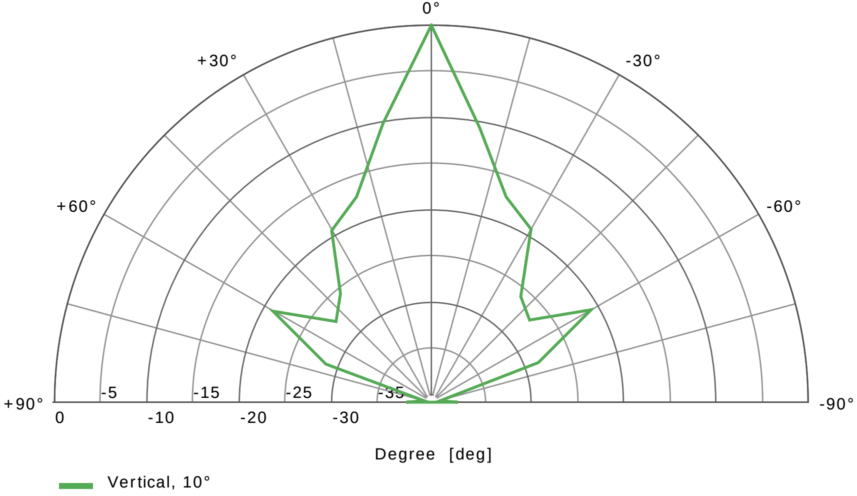
Directivity Measurements

Angular Resolution

The inherent directivity of the DUT will dictate the required angular resolution needed to accurately characterize its directivity in the measurement data.

"Spatial Aliasing"





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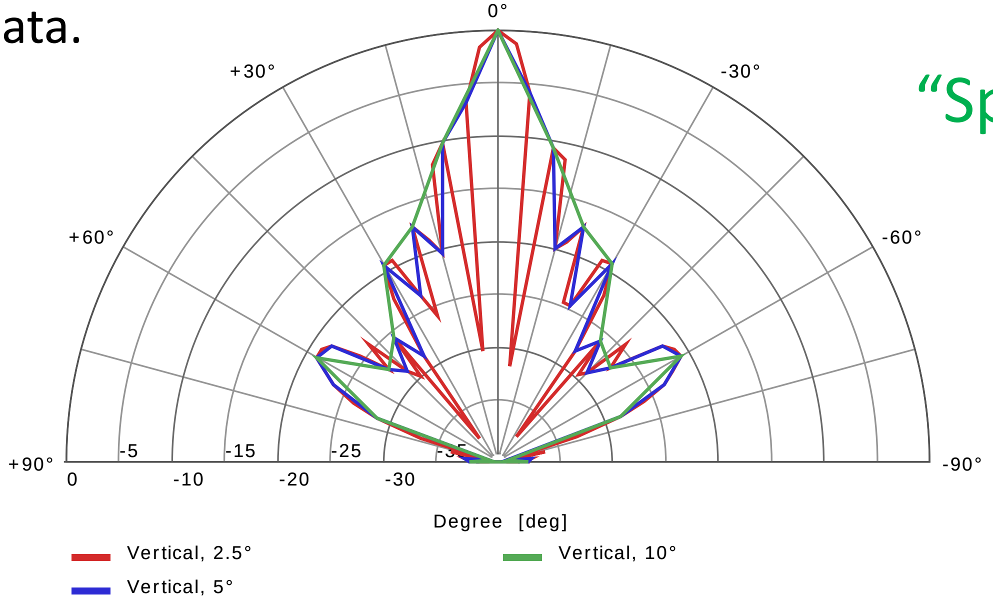


"Spatial Aliasing"

Directivity Measurements

Angular Resolution

Nulls at off-axis angles are related to the wavelength radiated and the dimension of the source in the plane of interest.

$$\theta_i = \sin^{-1} i \frac{c}{fl}$$

To avoid undersampling ("aliasing") a minimum resolution of 1/2 the angular value between the first and second nulls should be used.

$$\Delta\theta_{Crit} = \frac{1}{2}\left[\sin^{-1}\left(2\frac{c}{fl}\right) - \sin^{-1}\left(\frac{c}{fl}\right)\right]$$

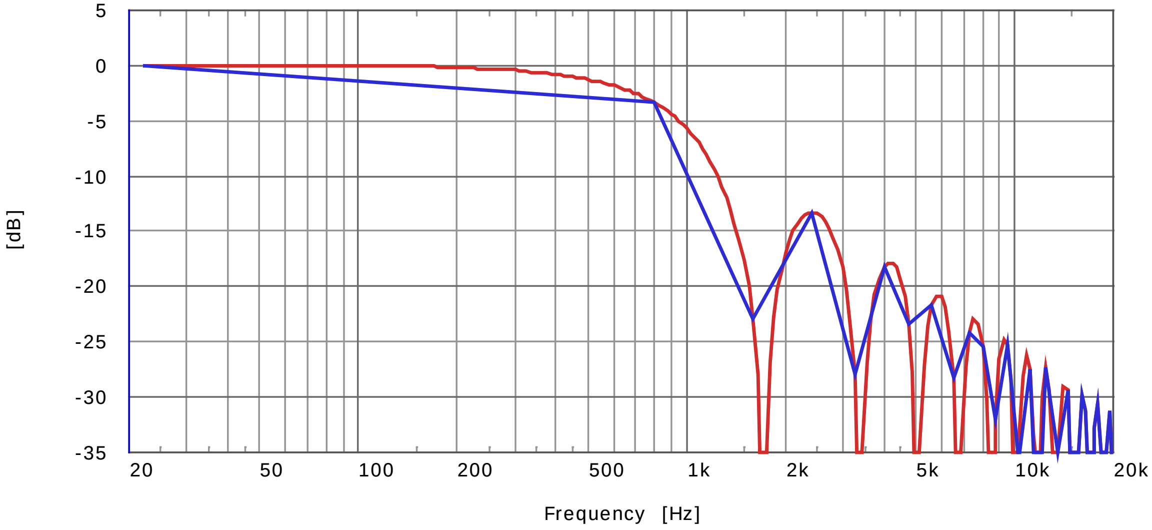
Assure that the on-axis measurement is truly on-axis. For highly directional sources a slight angular deviation for the on-axis reference yields an incorrect level, artificially increasing the level of the side lobes.

Feistel, et. al., "Methods and Limitations of Line Source Simulations", JAES, 2009 June

Directivity Measurements

Frequency Resolution

The inherent directivity & frequency response of the DUT will dictate the required frequency resolution needed to accurately characterize its response in the measurement data.



Directivity Measurements

Frequency Resolution

Nulls in the frequency response are related to the on-axis or off-axis angle and the dimension of the source.

$$f_i = i\frac{c}{l \sin\theta}$$

To avoid undersampling (“aliasing”) there should be at least 2 data points between adjacent nulls.

$$\Delta f_{crit} = \frac{c}{2l \sin\theta} \qquad \Delta f_{crit} = \frac{c}{2l}$$

The highest resolution requirement occurs for θ = 90°

Feistel, et. al., “Methods and Limitations of Line Source Simulations”, JAES, 2009 June

Directivity Measurements

Selection of POR (Point of Rotation)

The POR should be within a certain distance from the acoustical center of the source being measured. This distance must be evaluated in 3D, not just a single dimension.

One criterion for the maximum distance is based on the measurement distance, d, and the highest frequency of interest, f.

$$x_{Crit} = \sqrt{cd/_{4f}}$$

A second criterion for the maximum distance is based on the angular resolution, $\Delta\theta$, used for the directivity measurements and the highest frequency of interest, f.

$$x_{Crit} \approx \frac{c}{4f\sin(\Delta\theta)}$$

These are maximum upper limits. I often use a value 1/2 given by these.

Feistel, et. al., "Modeling of Loudspeaker Systems Using High Resolution Data", JAES, 2007 July/August

Directivity Measurements

Selection of POR (Point of Rotation)

The POR should be within a certain distance from the source being measured. This distance must be evaluated in 3D, not just a single dimension.

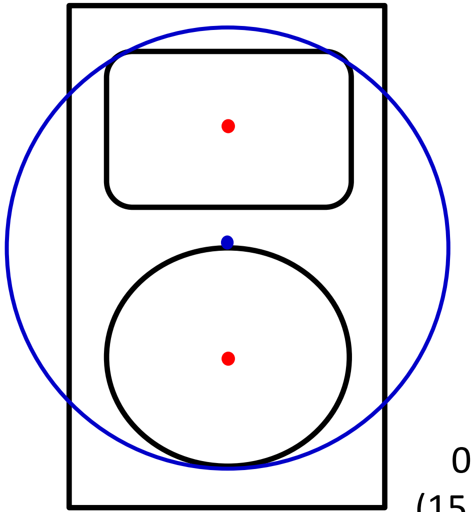
Sometimes the true location of the acoustical source can vary with frequency. This should be considered when choosing the POR.

Directivity Measurements

Selection of POR – Example (*for modeling only this loudspeaker*)

Measuring a two-way loudspeaker distance of 4 meters at 5° increments. The crossover frequency is approx. 1.2 kHz with 4th order filters (24 dB/oct).

f is about 1 octave higher than the crossover frequency, 2.4 kHz.



Radius of circle is 0.189 m

$$x_{Crit} = \frac{1}{2}\sqrt{cd/4f}$$

x_{Crit} = 0.189 m (7.44 in)

$$x_{Crit} \approx \frac{1}{2} * \frac{c}{4f\sin(\Delta\theta)}$$

$x_{Crit} \approx$ 0.206 m (16.2 in)

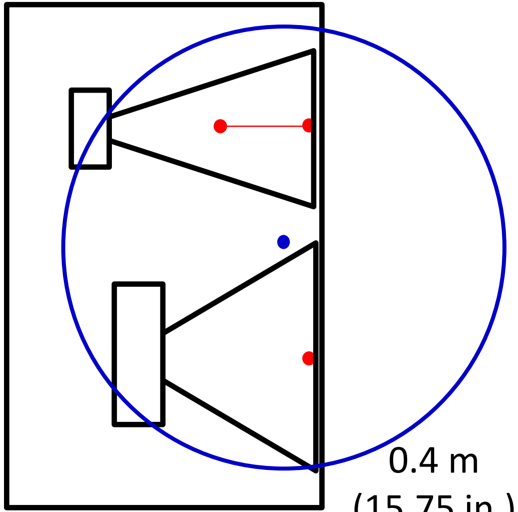
Directivity Measurements

Selection of POR – Example (*for modeling only this loudspeaker*)

Measuring a two-way loudspeaker distance of 4 meters at 5° increments. The crossover frequency is approx. 1.2 kHz with 4th order filters (24 dB/oct).

f is about 1 octave higher than the crossover frequency, 2.4 kHz.

0.7 m
(27.5 in.)



0.4 m
(15.75 in.)

Radius of circle is 0.189 m

$$x_{Crit} = \frac{1}{2}\sqrt{cd/4f}$$

$$x_{Crit} = 0.189 \text{ m } (7.44 \text{ in})$$

$$x_{Crit} \approx \frac{1}{2} * \frac{c}{4f\sin(\Delta\theta)}$$

$$x_{Crit} \approx 0.206 \text{ m } (16.2 \text{ in})$$

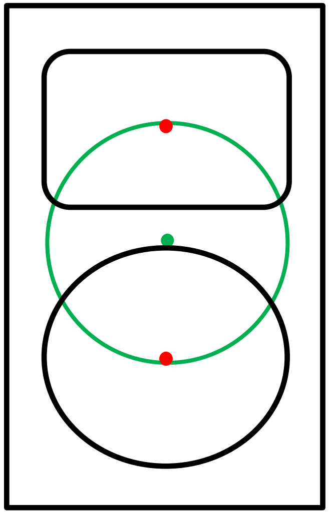
Directivity Measurements

Selection of POR – Example (*for modeling only this loudspeaker*)

Measuring a two-way loudspeaker distance of 4 meters at 5° increments. The crossover frequency is approx. 1.2 kHz with 2nd order filters (12 dB/oct).

f is about 2 octaves higher than the crossover frequency, 4.8 kHz.

0.7 m
(27.5 in.)



0.4 m
(15.75 in.)

Radius of circle is 0.103 m

$$x_{Crit} = \frac{1}{2}\sqrt{{cd}/{4f}}$$

$$x_{Crit} = 0.134 \text{ m } (5.28 \text{ in})$$

$$x_{Crit} \approx \frac{1}{2} * \frac{c}{4f\sin(\Delta\theta)}$$

$$x_{Crit} \approx 0.103 \text{ m } (4.05 \text{ in})$$

Directivity Measurements

Selection of POR – Example (*for modeling only this loudspeaker*)

Measuring a two-way loudspeaker distance of 4 meters at 5° increments. The crossover frequency is approx. 1.2 kHz with 2nd order filters (12 dB/oct).

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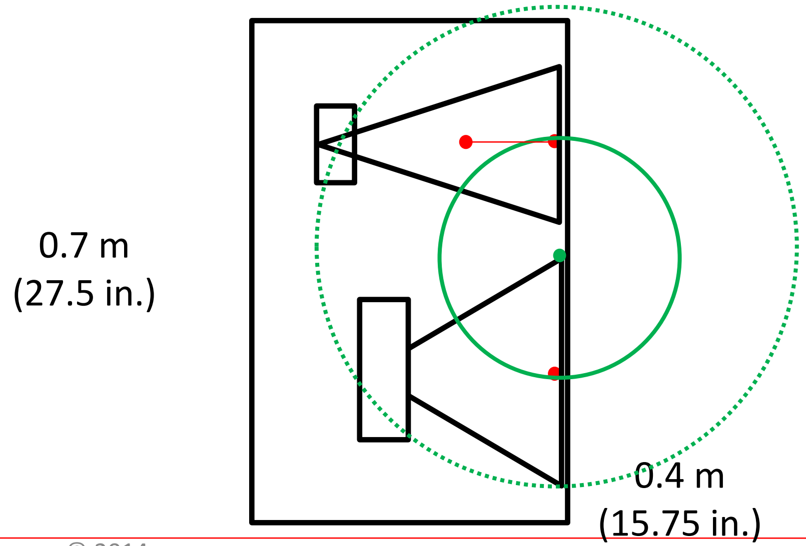


Radius of circle is 0.103 m

$$x_{Crit} = \frac{1}{2}\sqrt{{cd}/{4f}}$$

$$x_{Crit} = 0.134\text{ m }(5.28\text{ in})$$

$$x_{Crit} \approx \frac{1}{2} * \frac{c}{4f\sin(\Delta\theta)}$$

$$x_{Crit} \approx 0.103\text{ m }(4.05\text{ in})$$

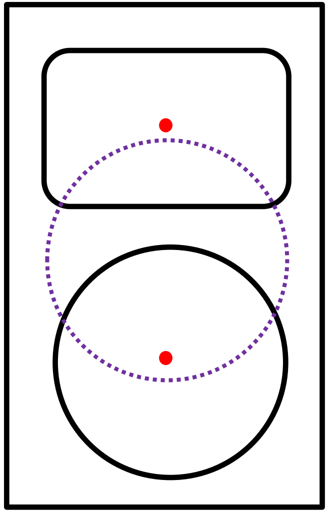
Directivity Measurements

Selection of POR – Example (*for modeling this loudspeaker with others*)

We want to be able to model arrays/clusters up to about 10 kHz. Calculate values for x_{Crit} separately for the HF & LF.

f_{LF} is the highest frequency from the LF pass band, about 4.8 kHz.

0.7 m
(27.5 in.)



0.4 m
(15.75 in.)

Diameter of circle for LF is 0.103 m & contains center of LF

$$x_{Crit} = \frac{1}{2}\sqrt{{cd}/{4f}}$$

$$x_{Crit} = 0.134 \text{ m } (5.28 \text{ in})$$

$$x_{Crit} \approx \frac{1}{2} * \frac{c}{4f\sin(\Delta\theta)}$$

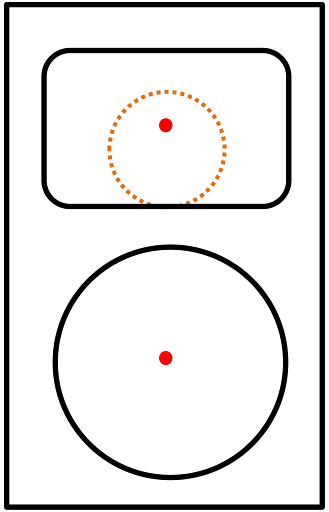
$$x_{Crit} \approx 0.103 \text{ m } (4.05 \text{ in})$$

Directivity Measurements

Selection of POR – Example (*for modeling this loudspeaker with others*)

We want to be able to model arrays/clusters up to about 10 kHz. Calculate values for x_{Crit} separately for the HF & LF.

f_{HF} is the highest frequency from the HF pass band or limit of modeling, about 10 kHz.



Diameter of circle for HF is 0.049 m & contains center of HF

$$x_{Crit} = \frac{1}{2}\sqrt{{cd}/{4f}}$$

x_{Crit} = 0.093 m (3.65 in)

$$x_{Crit} \approx \frac{1}{2} * \frac{c}{4f\sin(\Delta\theta)}$$

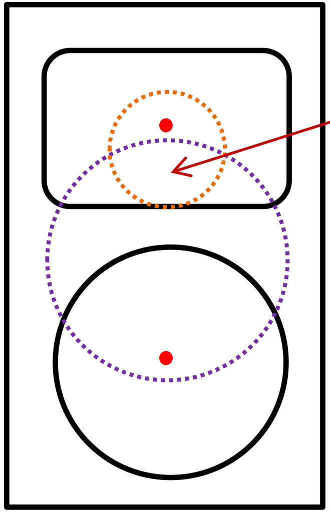
$x_{Crit} \approx$ 0.049 m (1.94 in)

Directivity Measurements

Selection of POR – Example (*for modeling this loudspeaker with others*)

We want to be able to model arrays/clusters up to about 10 kHz. Calculate values for x_{Crit} separately for the HF & LF.

0.7 m
(27.5 in.)



0.4 m
(15.75 in.)

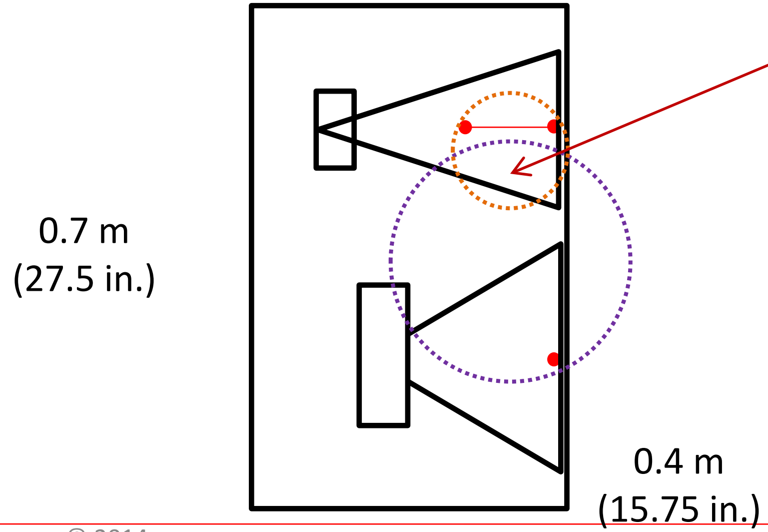
The POR can be placed in the region where the two circles overlap.

These circles do not include the 1/2 value showed earlier so these are is the maximum limits for the location of the POR. It must be within this area. The closer to the center of the overlap region is better.

Directivity Measurements

Selection of POR – Example (*for modeling this loudspeaker with others*)

We want to be able to model arrays/clusters up to about 10 kHz. Calculate values for x_{Crit} separately for the HF & LF.



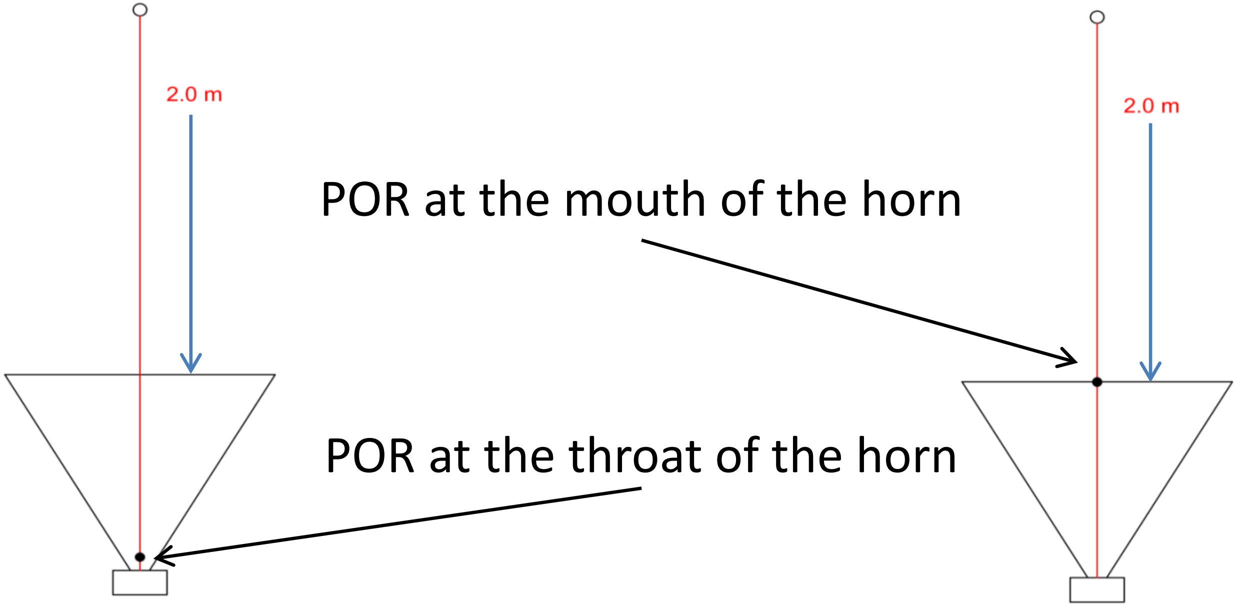
The POR can be placed in the region where the two circles overlap.

These circles do not include the 1/2 value showed earlier so these are is the maximum limits for the location of the POR. It must be within this area. The closer to the center of the overlap region is better.

Directivity Measurements

Change in Arrival Time of IR

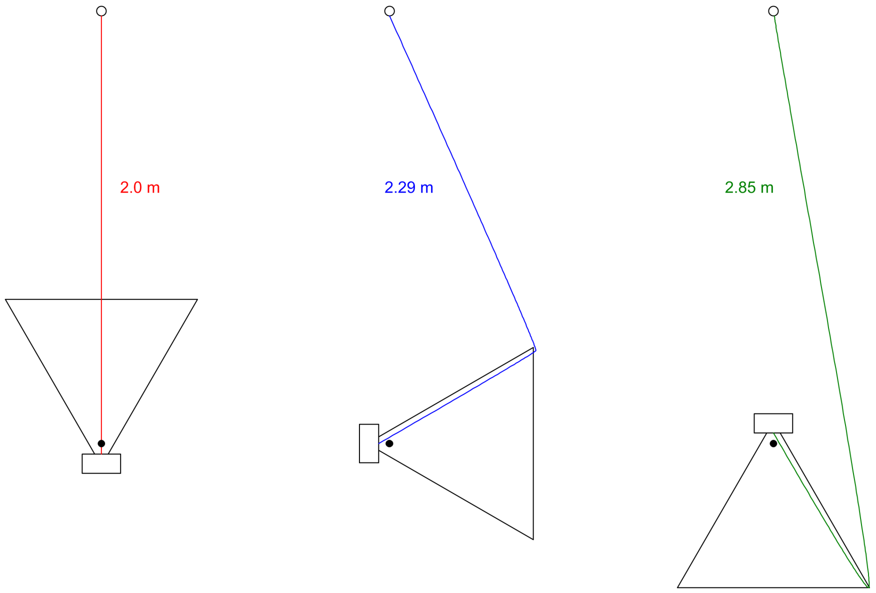
The arrival time of the IR can change as the DUT is rotated about its POR. This must be taken into account when defining the IR window.



Directivity Measurements

Change in Arrival Time of IR – POR at Throat

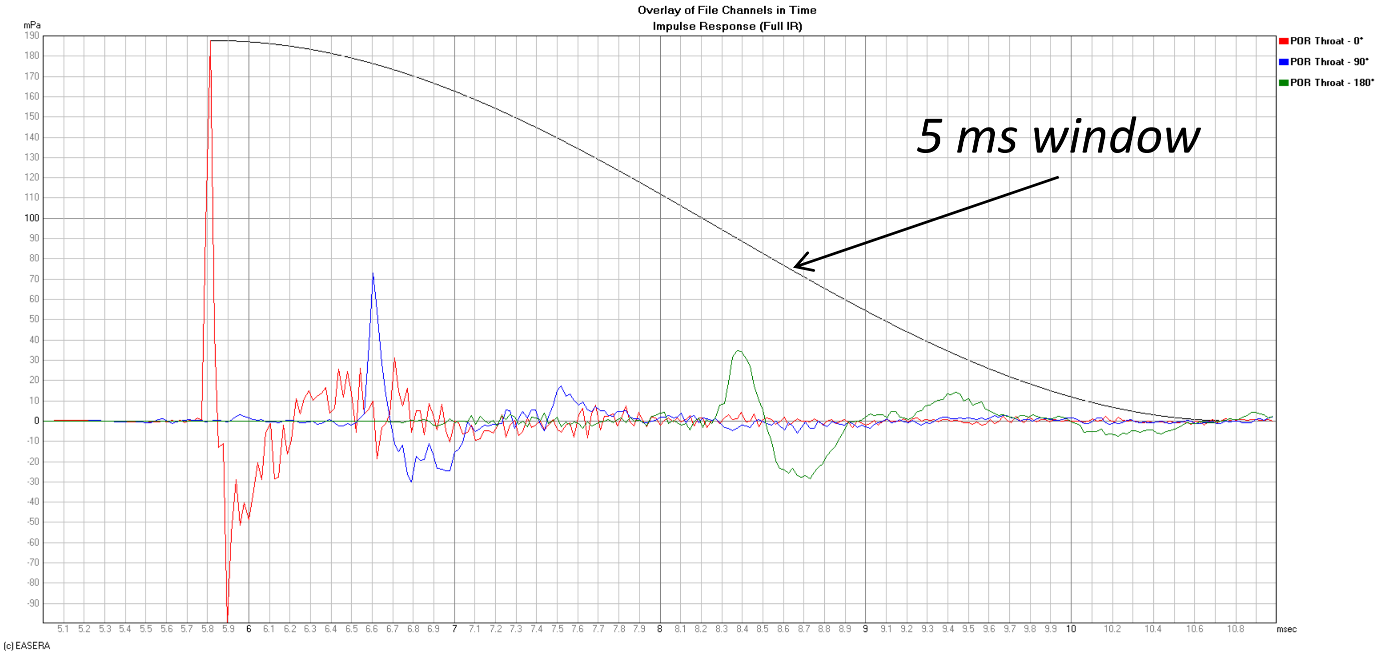
The distance from the mouth to the microphone changes greatly as the DUT is rotated.



Directivity Measurements

Change in Arrival Time of IR – POR at Throat

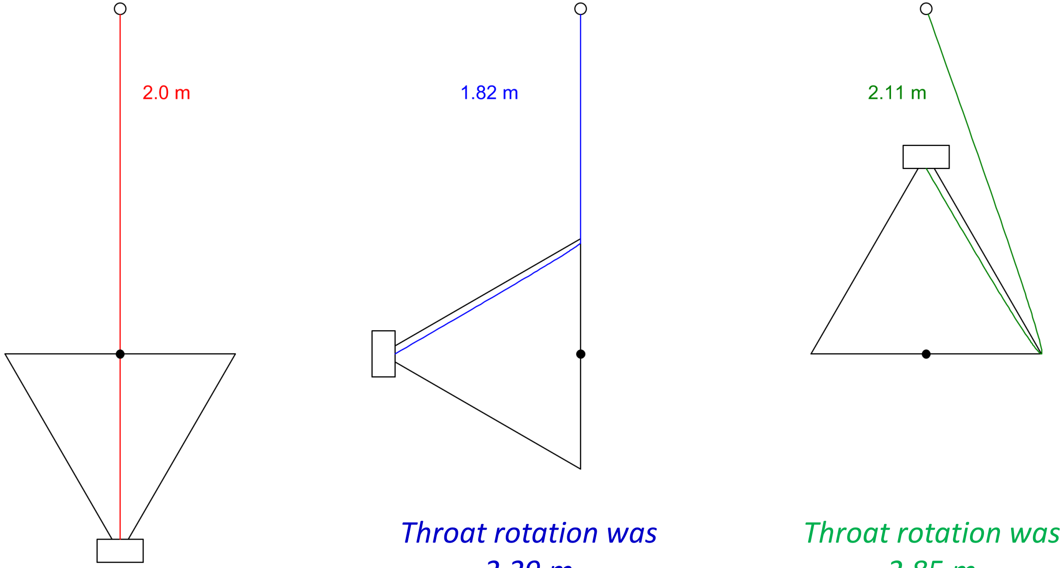
The arrival time of the IR at the microphone changes greatly as the DUT is rotated



Directivity Measurements

Change in Arrival Time of IR – POR at Mouth

The distance from the mouth to the microphone changes less as the DUT is rotated.



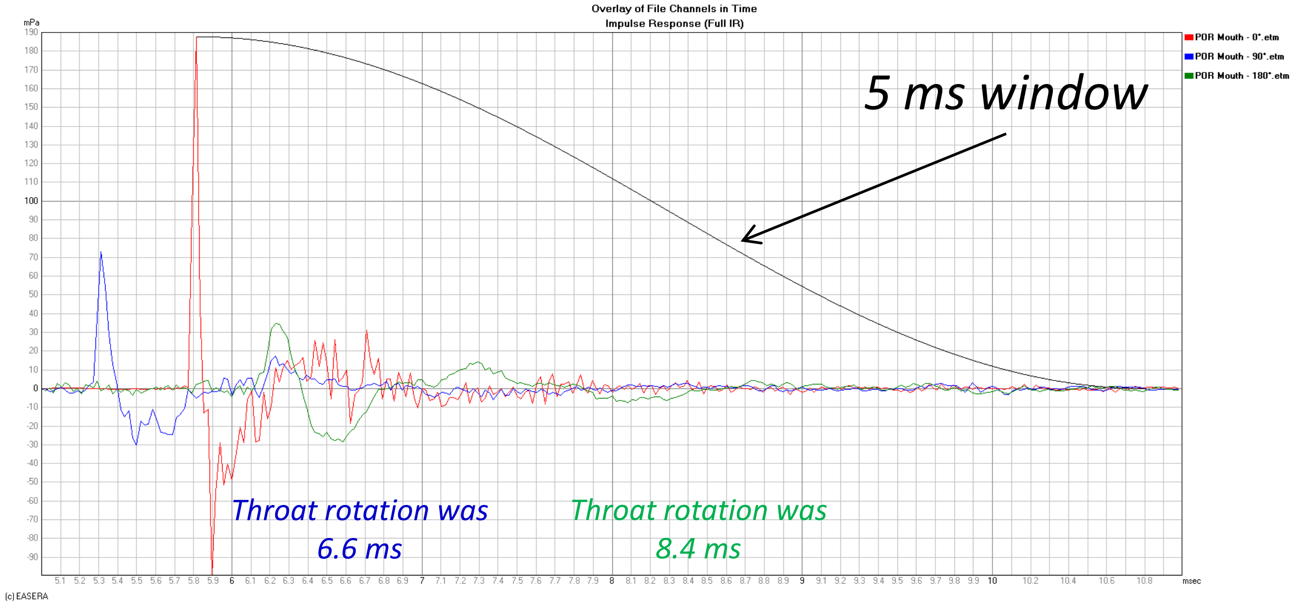
Throat rotation was 2.29 m

Throat rotation was 2.85 m

Directivity Measurements

Change in Arrival Time of IR – POR at Mouth

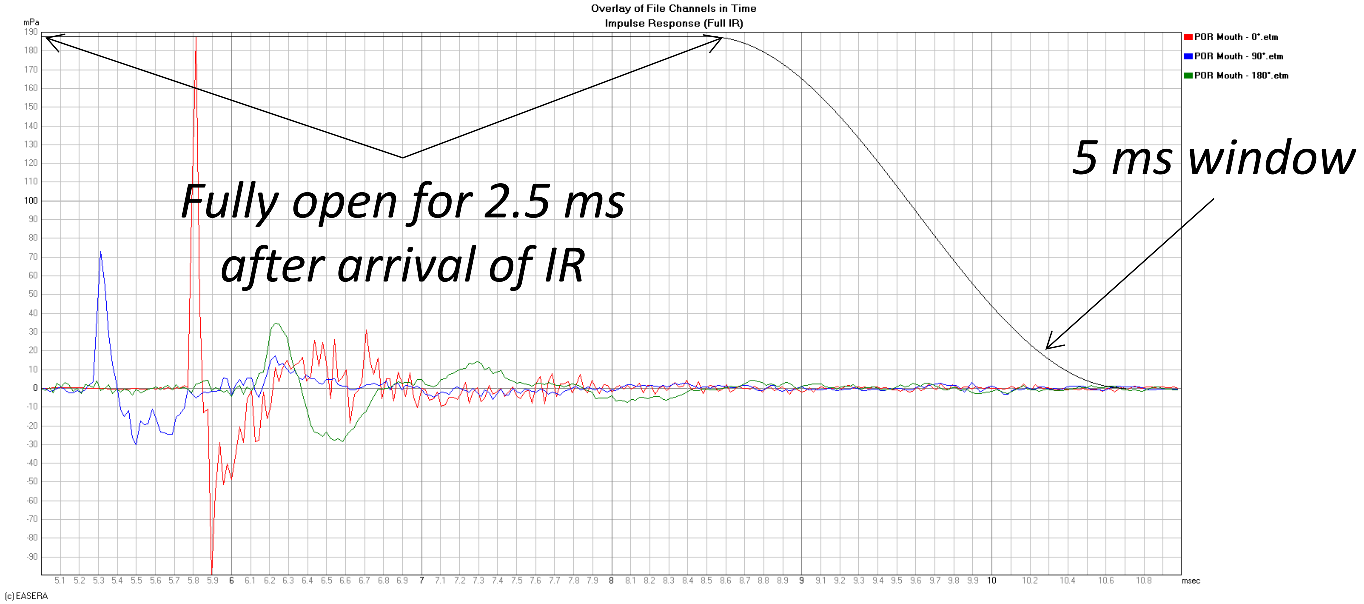
The arrival time of the IR at the microphone changes as the DUT is rotated, but not as much as when rotated about the throat.



Directivity Measurements

Change in Arrival Time of IR – Better Windowing

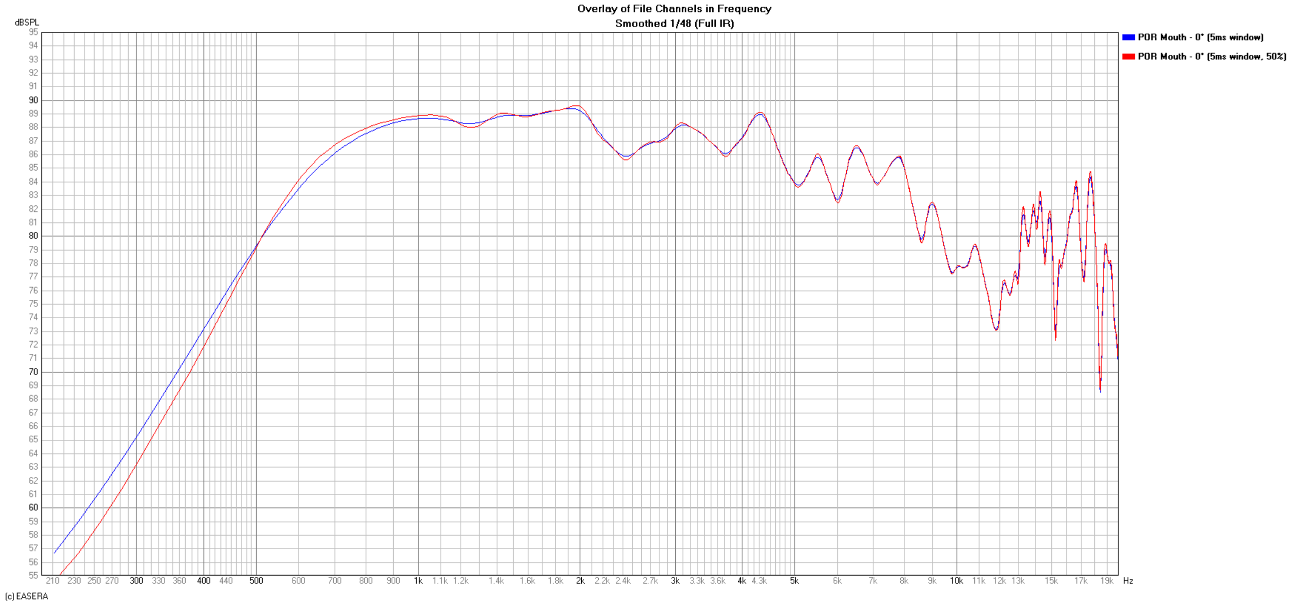
Start the window about half-way from the arrival of the on-axis IR to when the window needs to be fully closed. Allows for IR "walking" around before the window starts to close.



Directivity Measurements

Change in Arrival Time of IR – Better Windowing

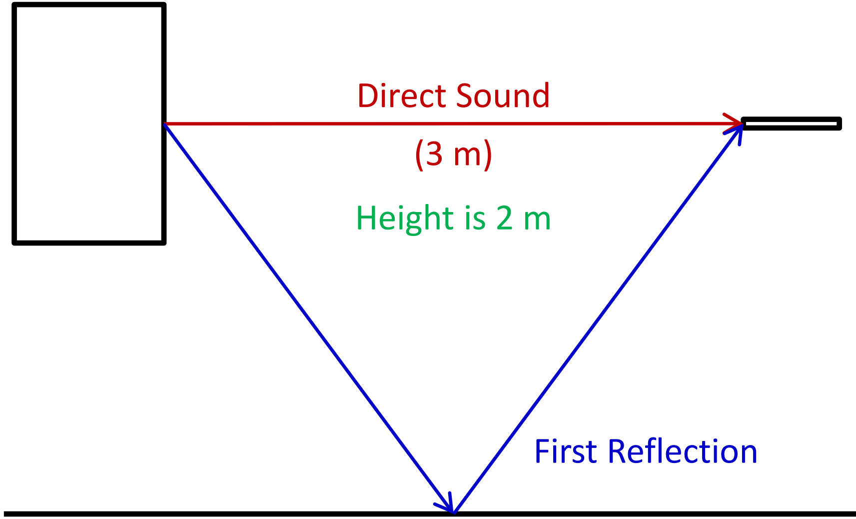
Also yields better frequency resolution as less energy is attenuated in the time domain since the window is fully open longer.

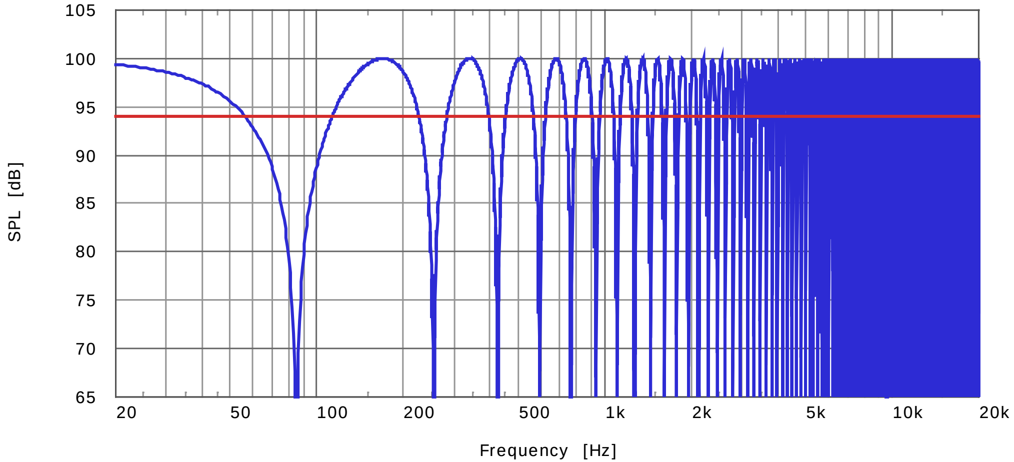


Ground Plane Measurements

The "Ground Bounce"

The difference in path length between the direct sound & the reflected sound causes comb filtering in the frequency response at the mic location.

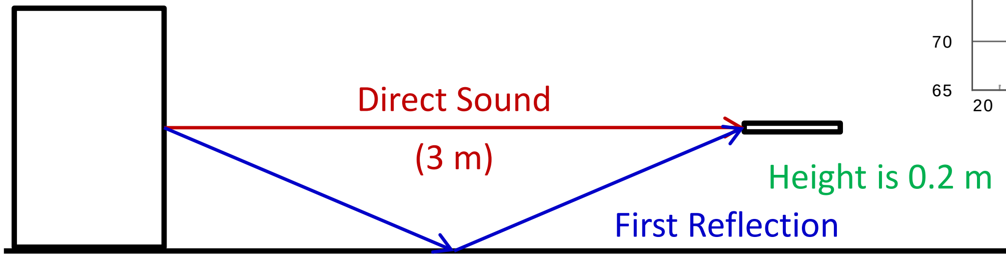


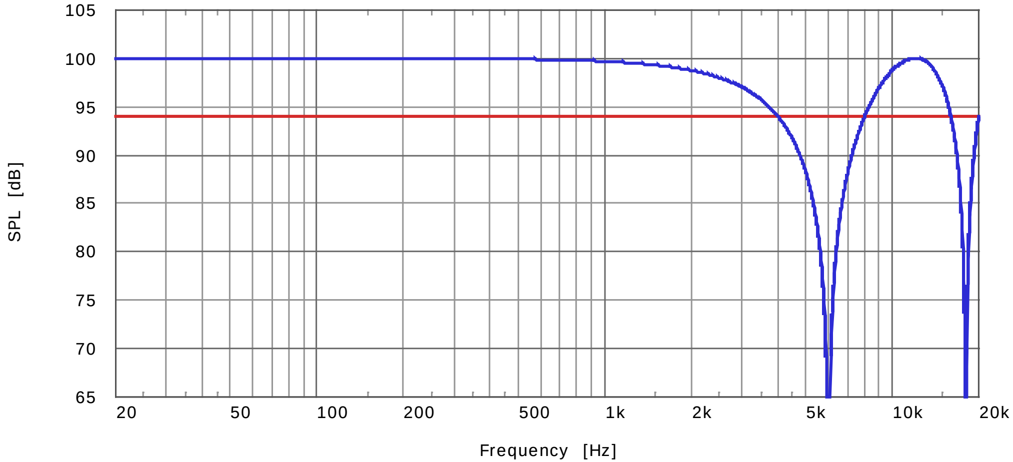


Ground Plane Measurements

The "Ground Bounce"

Decreasing the height above the ground decreases the path length difference.
This increases the frequency of the first notch in the comb filter response.

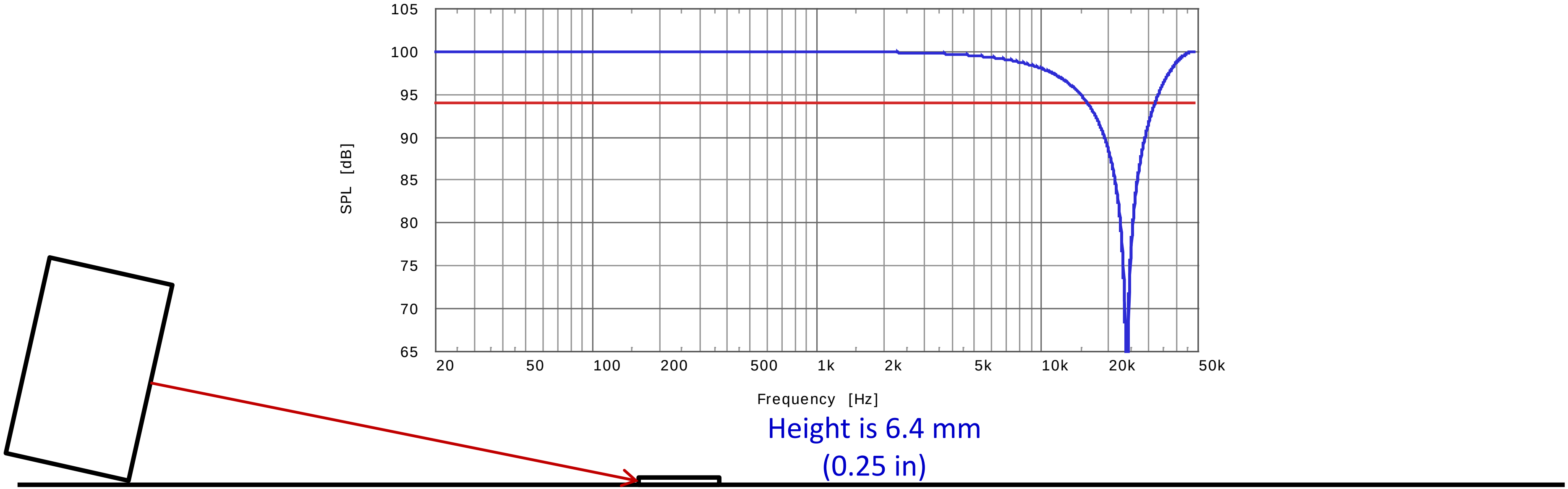




Ground Plane Measurements

Using the "Ground Bounce" Advantageously

Placing a 0.5 inch mic directly on perfectly reflective surface.



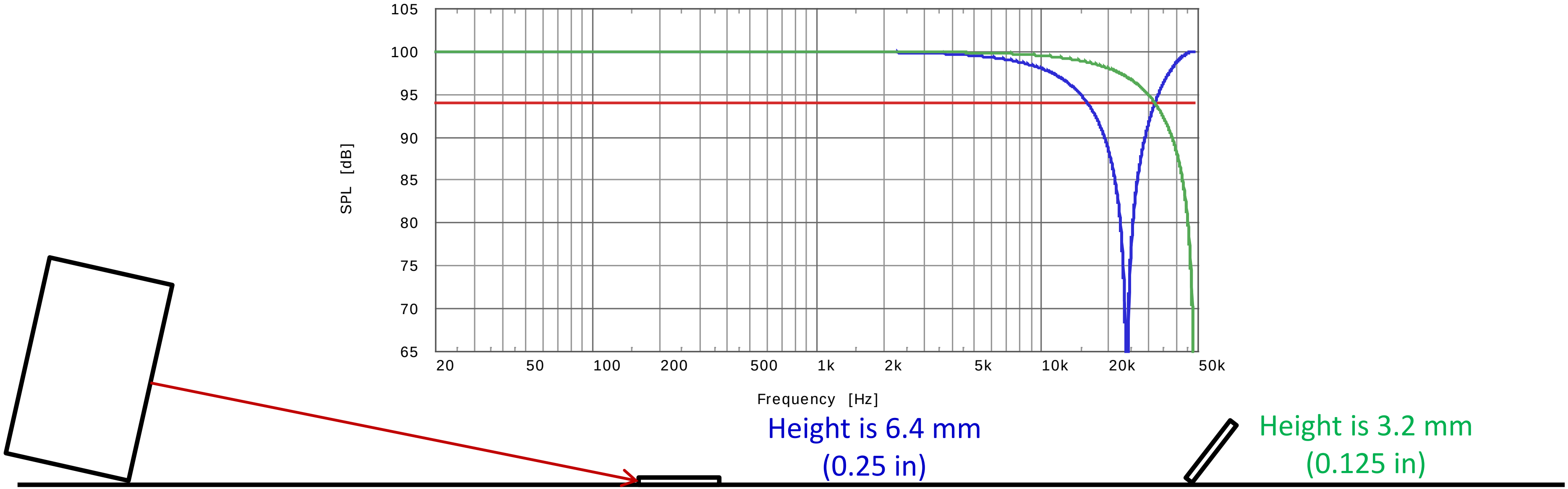
Height is 6.4 mm
(0.25 in)

Ground Plane Measurements

Using the "Ground Bounce" Advantageously

Placing a 0.5 inch mic directly on perfectly reflective surface.

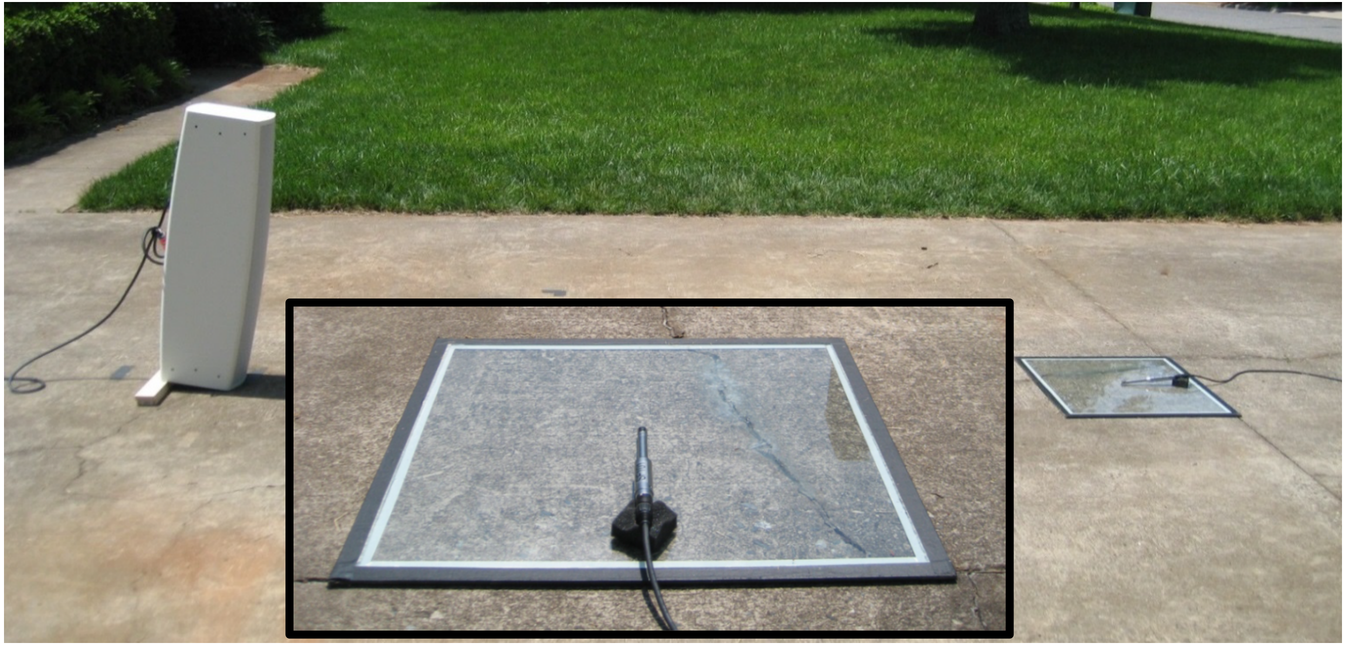
Angling 0.5 inch mic down or using 0.25 inch mic.



Ground Plane Measurements

Ground Plane Reflectivity

Not all surfaces are equally reflective at all frequencies.



Impedance Measurements

Dual-Channel FFT Measurement System

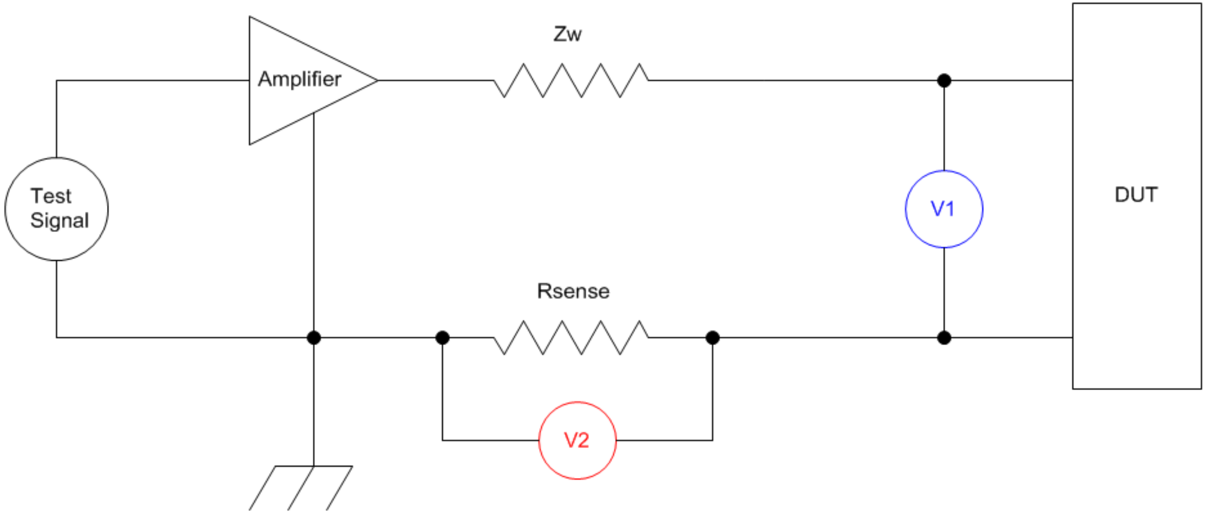
Transfer Function Type Measurement

Measurement channel
V1 = V (voltage)
(voltage across the DUT)

Reference channel
V2 ∝ I (current)
(voltage across the sense resistor is proportional to current through the DUT)

TF = V1 / V2

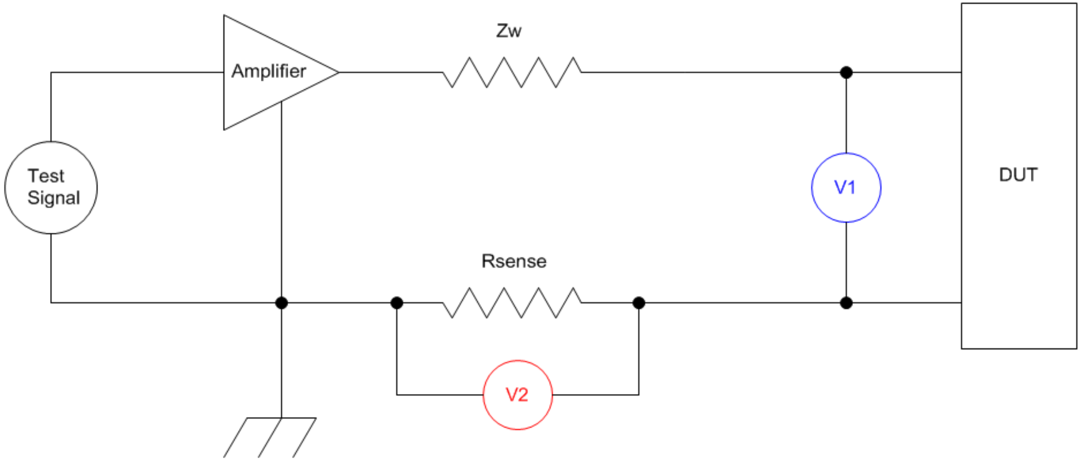
Z = V/I



Impedance Measurements

Constant Current Method

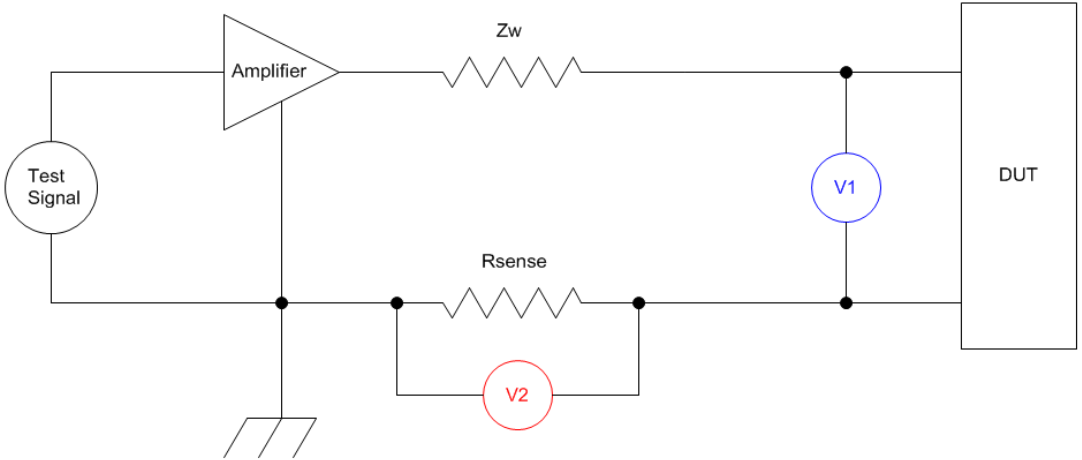
1. R_{Sense} is relatively ***large*** compared to the impedance of the DUT (e.g. ≥ 1 kΩ).
2. Often used for small signal measurements (e.g. Thiele-Small parameters).
3. Large voltage drop across R_{Sense} so very low excitation voltage across DUT.
4. Susceptible to acoustical noise at DUT contaminating the measurement.



Impedance Measurements

Constant Voltage Method

1. R_{Sense} is relatively ***small*** compared to the impedance of the DUT (e.g. ≤ 1 Ω).
2. Often used for large signal measurements.
3. Small voltage drop across R_{Sense} large excitation voltage across DUT possible.
4. Less susceptible to acoustical noise at DUT contaminating the measurement



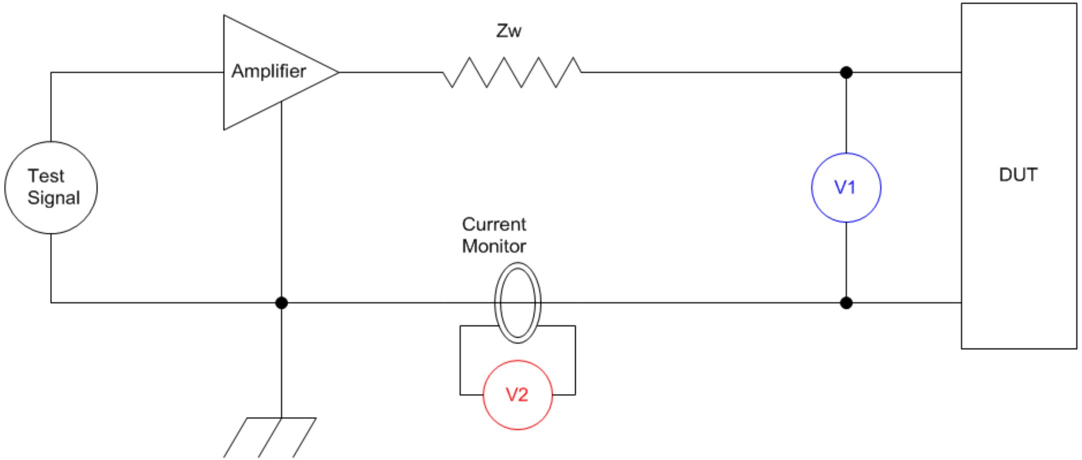
Impedance Measurements

Constant Voltage Method

Impedance Measurements

Constant Voltage Method

1. Current is monitored ***magnetically***. No R_{Sense} in the connection to DUT.
2. Can be used for large and small signal measurements.
3. Very large excitation voltage across DUT possible.
4. Less susceptible to acoustical noise at DUT contaminating the measurement.



Impedance Measurements

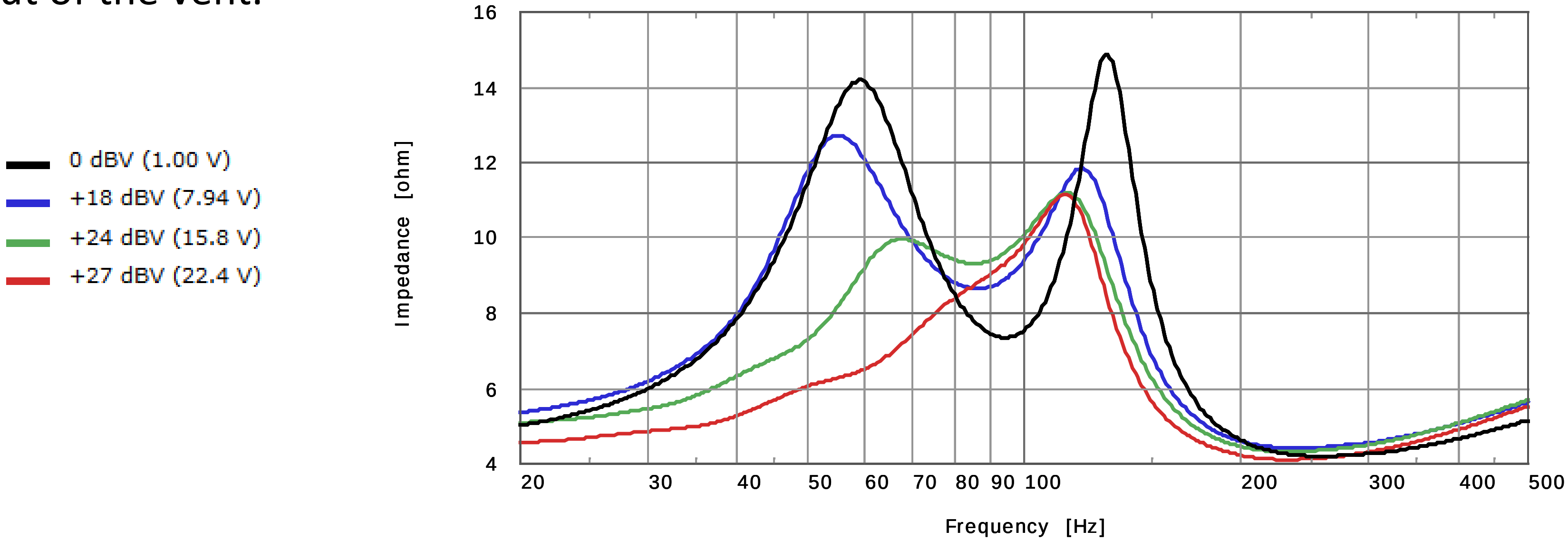
Constant Voltage Method

Pearson 411 Current Monitor

Impedance Measurements

Vent Saturation

At high input signal levels with a lot of low frequency content loudspeaker vents can become “saturated” due to high particle velocity of the air attempting to moving in & out of the vent.



Maximum Input Voltage

Helps to more accurately determines maximum SPL in real use than traditional “power handling”

Codified in AES2-2012 and ANSI/CEA-2034 standards

Based on real-time change in the frequency response of the DUT as the input level in slowly increased

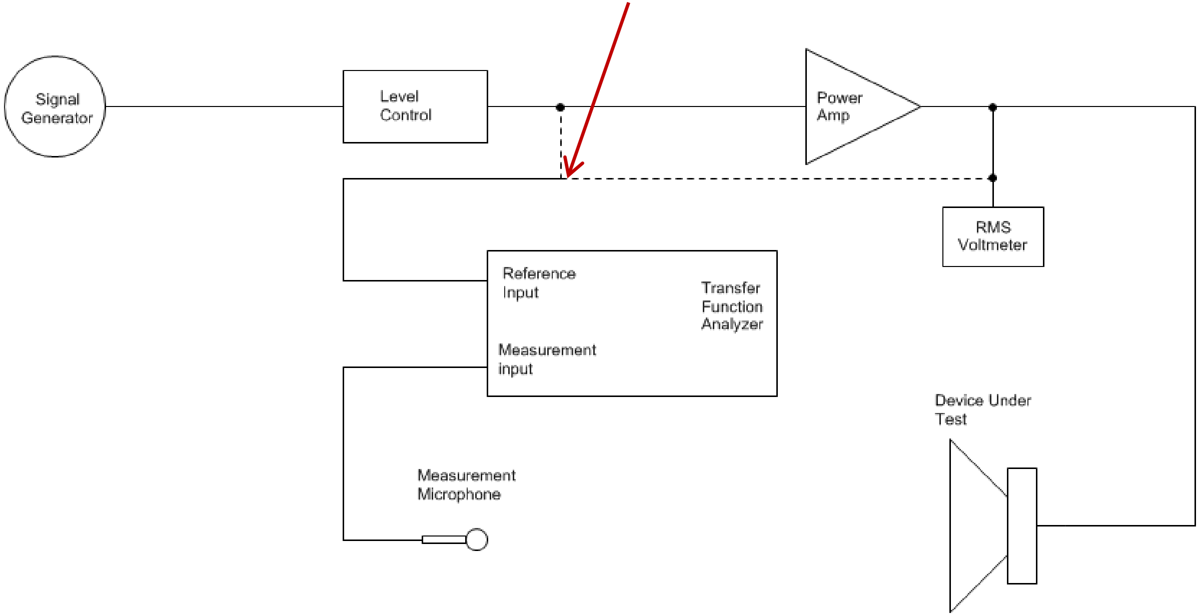
The MIV is determined when the frequency response of the DUT has change by 3 dB compared to it’s frequency response at a low input level.

Test method was originally developed by Pat Brown at Synergetic Audio Concepts

Maximum Input Voltage

Measurement Setup

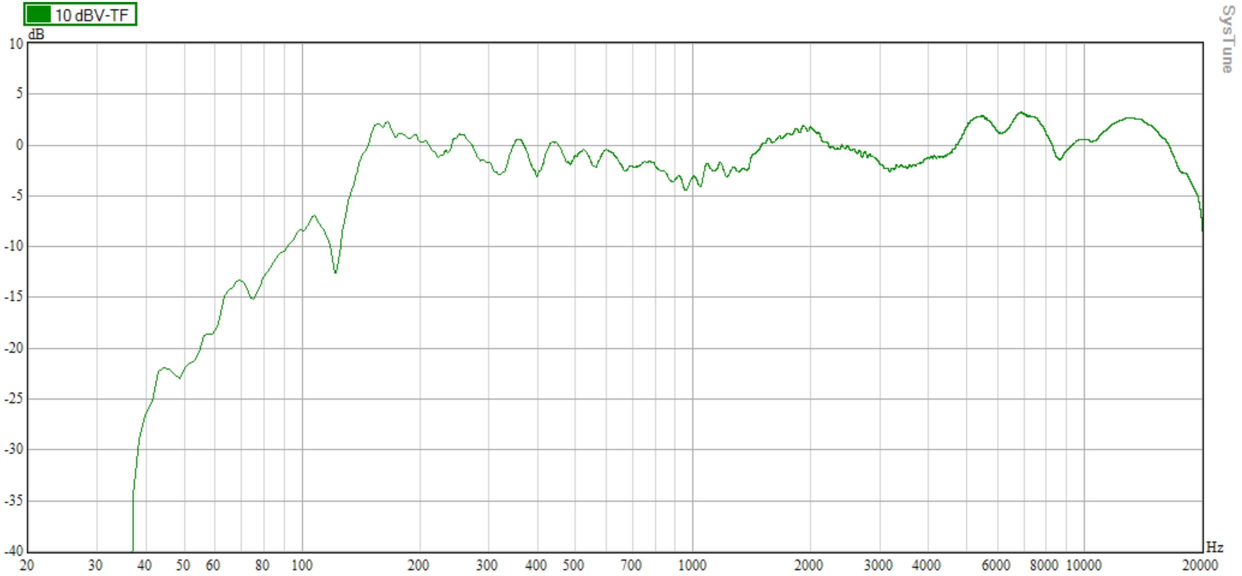
The reference input to the dual-channel FFT is after the level control. Only non-linear changes in the output level ("compression") of the DUT are recorded.



Maximum Input Voltage

Normalization to Initial Measurement

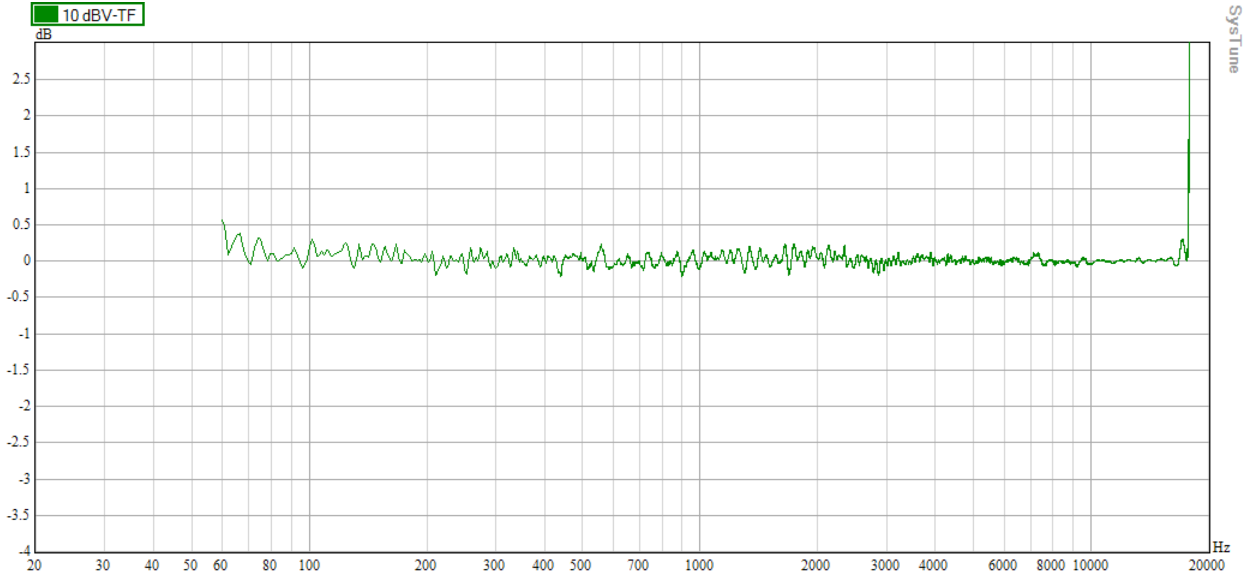
The initial measurement at a low input level (0 dBV or 10 dBV) is used to normalize all of the measurements.



Maximum Input Voltage

Normalization to Initial Measurement

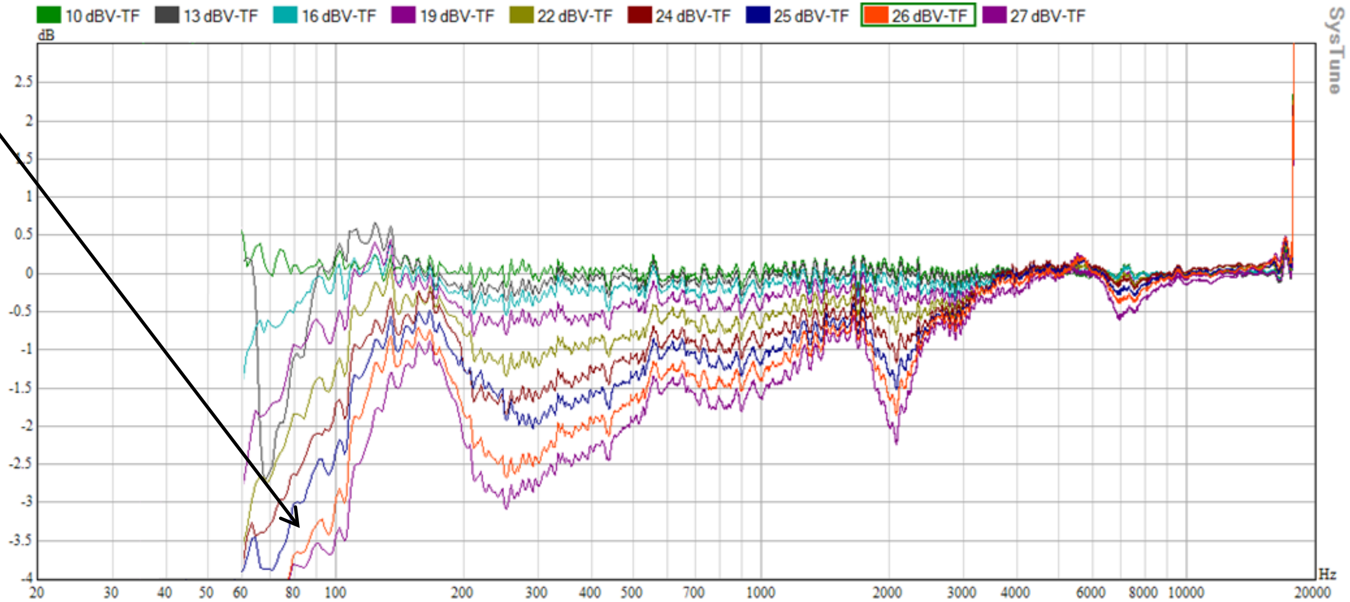
The initial measurement at a low input level (0 dBV or 10 dBV) is used to normalize all of the measurements.



Maximum Input Voltage

Typical Changes During Testing

Two-way ceiling loudspeaker, LF limit 85 Hz, high pass filtered at 60 Hz
MIV determined by response change between 3.0 to 3.5 dB (85 Hz)

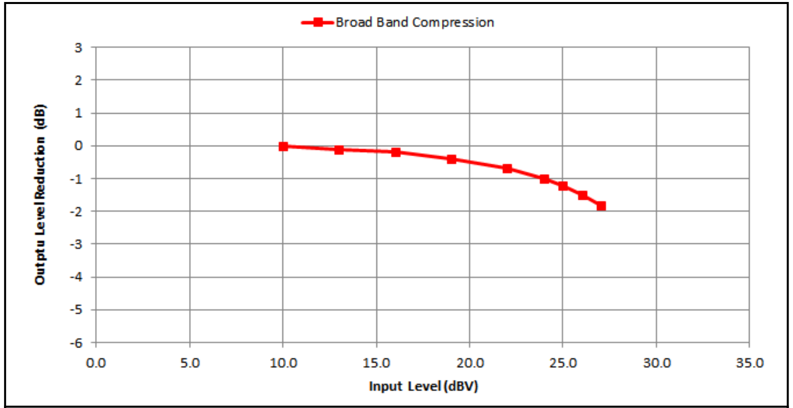


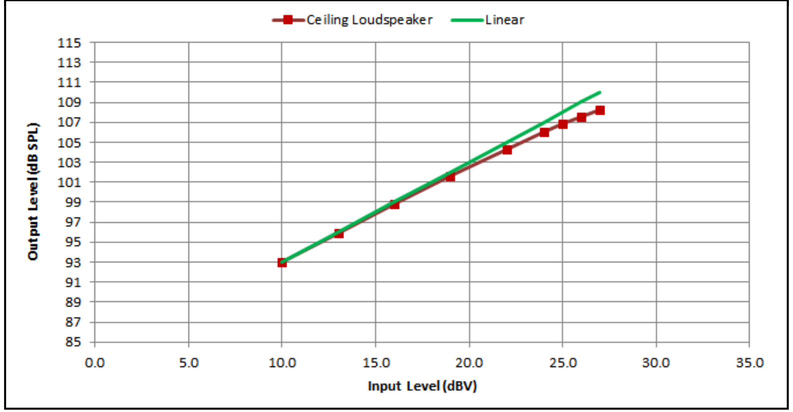
Maximum Input Voltage

Data Logged During Testing

Input voltage and output SPL
Ceiling Loudspeaker

		Distance (m)	Ground Plane	
		2.0	-6.0	
Input Level (dBV)	Voltage	dB SPL (2 m)	dB SPL (ref. 1 m)	Broad Band Compression
10.0	3.16	93.0	93.0	-
13.0	4.47	95.9	95.9	-0.1
16.0	6.31	98.8	98.8	-0.2
19.0	8.91	101.6	101.6	-0.4
22.0	12.59	104.3	104.3	-0.7
24.0	15.85	106.0	106.0	-1.0
25.0	17.78	106.8	106.8	-1.2
26.0	19.95	107.5	107.5	-1.5
27.0	22.39	108.2	108.2	-1.8



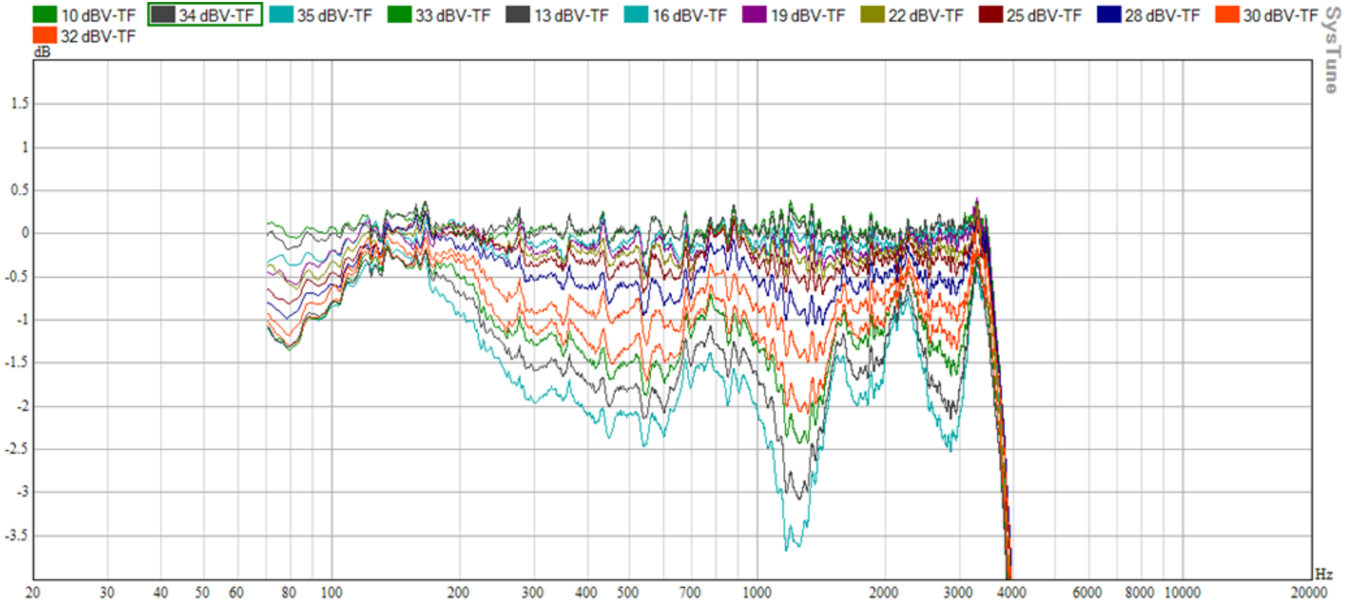


Maximum Input Voltage

Typical Changes During Testing

LF only of line array cabinet (dual 8 inch), band pass filtered 70 Hz to 3 kHz

MIV determined by response change between 3.0 to 3.5 dB (1.1 kHz)

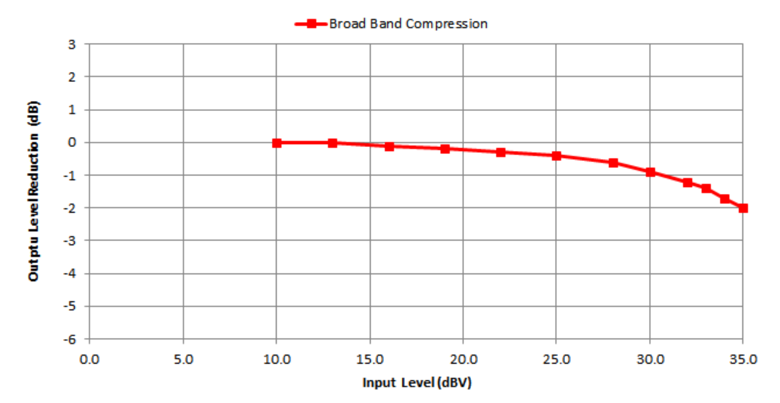


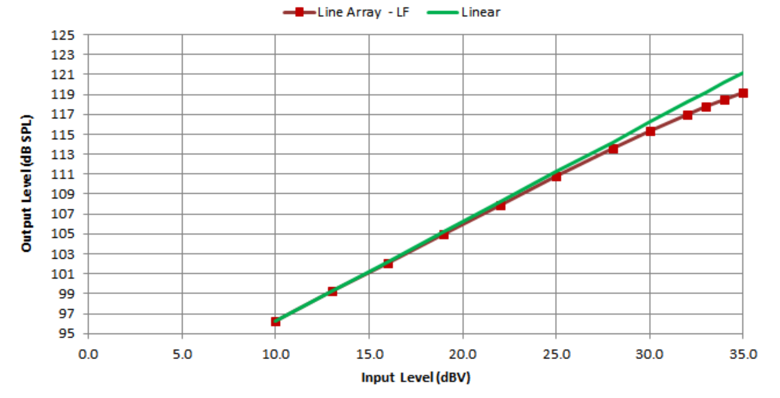
Maximum Input Voltage

Data Logged During Testing

Input voltage and output SPL
Line Array LF Section

		Distance (m)	Ground Plane	
		2.0	-6.0	
Input Level (dBV)	Voltage	dB SPL (2 m)	dB SPL (ref. 1 m)	Broad Band Compression
10.0	3.16	96.2	96.2	-
13.0	4.47	99.2	99.2	0.0
16.0	6.31	102.1	102.1	-0.1
19.0	8.91	105.0	105.0	-0.2
22.0	12.59	107.9	107.9	-0.3
25.0	17.78	110.8	110.8	-0.4
28.0	25.12	113.6	113.6	-0.6
30.0	31.62	115.3	115.3	-0.9
32.0	39.81	117.0	117.0	-1.2
33.0	44.67	117.8	117.8	-1.4
34.0	50.12	118.5	118.5	-1.7
35.0	56.23	119.2	119.2	-2.0



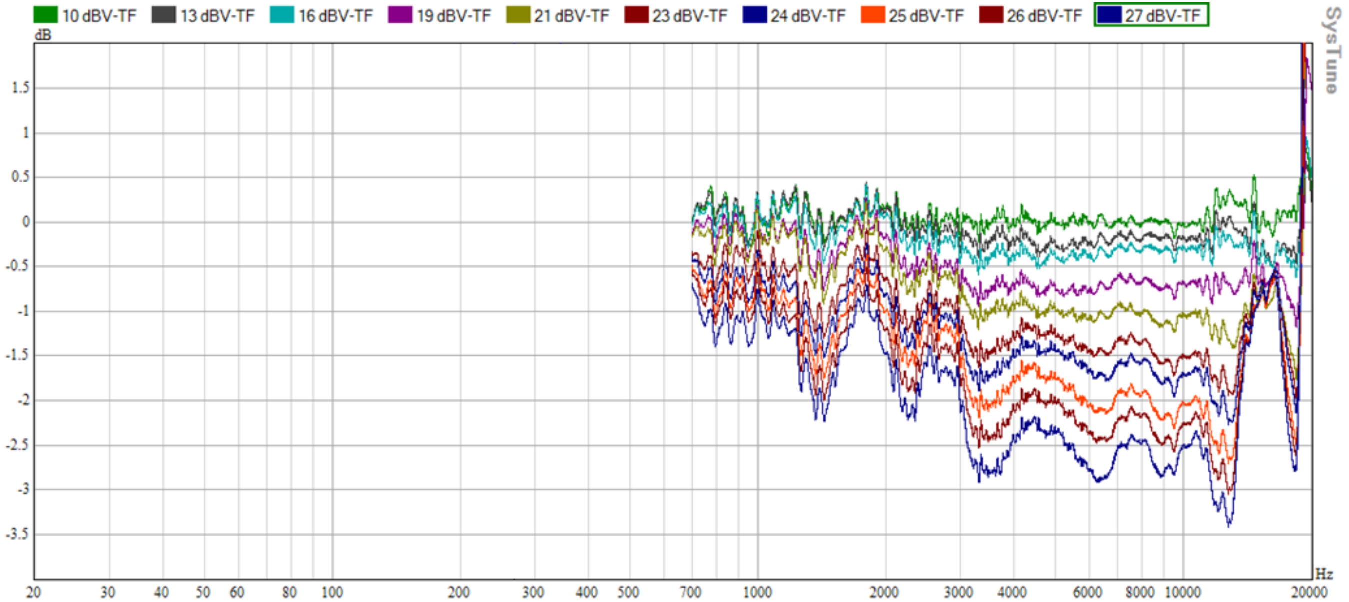


Maximum Input Voltage

Typical Changes During Testing

HF only of line array cabinet, high pass filtered 1 kHz

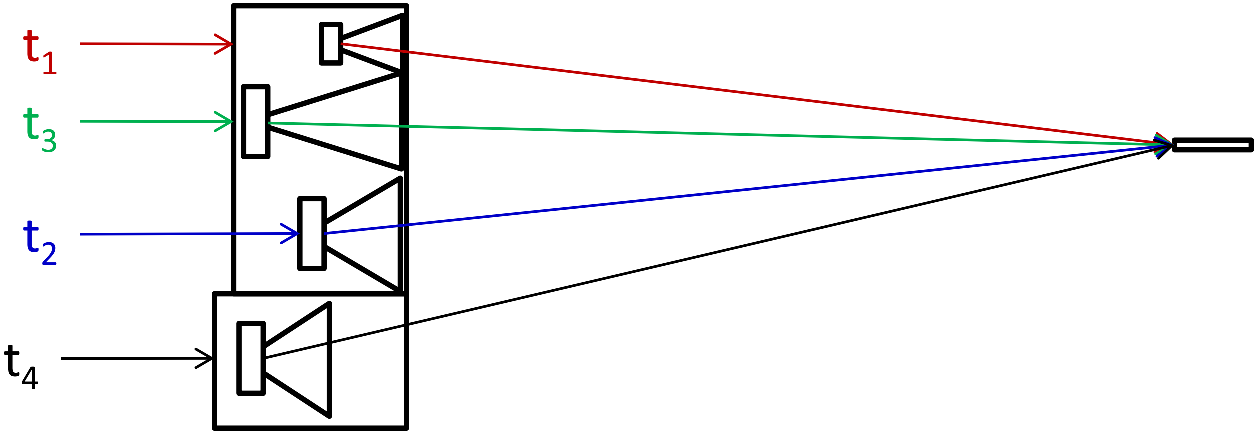
MIV determined by response change between 3.0 to 3.5 dB (12 kHz)



Synchronization (Alignment) of Pass Bands

Subjective Sound Quality

It is generally accepted that the perceived sound quality of a loudspeaker is improved when the signals radiated by different pass band (i.e. Sub, LF, MF, and HF) are synchronized; they arrive at the listener’s ears (or measurement mic) at the same time.

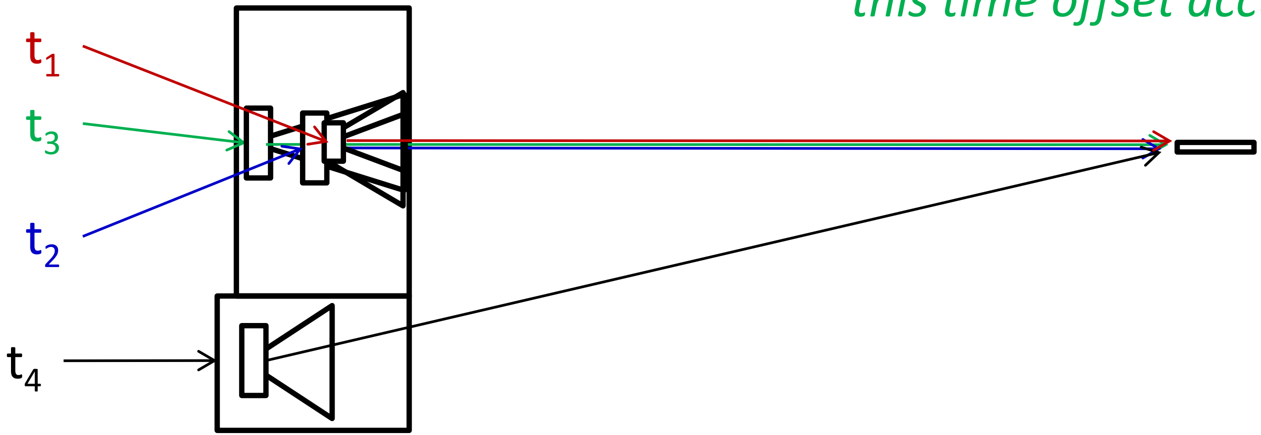


Synchronization (Alignment) of Pass Bands

Subjective Sound Quality

Even if we arrange the pass bands co-axially within the same enclosure, we still have an axial distance offset, and the resulting time offset between the pass bands.

How do we measure & determine this time offset accurately?



Synchronization (Alignment) of Pass Bands

Measuring Arrival Time

Impulse Response (time domain)

Can be very accurate

Must look at the initial arrival of energy, not energy peak

Phase Response & Group Delay (frequency domain)

Limited to the LF limit and frequency resolution of the windowed data.

Measurements requiring small frequency resolution indoors can be problematic.

Limitations to accuracy of arrival time when using in-band phase response

Must bypass all crossover and EQ filters to accurately determine arrival of DUT

Must look at the value of phase or group delay in the HF limit of the DUT for accuracy

This may be in the stop band of the DUT. Extremely good S/N is required.

Synchronization (Alignment) of Pass Bands

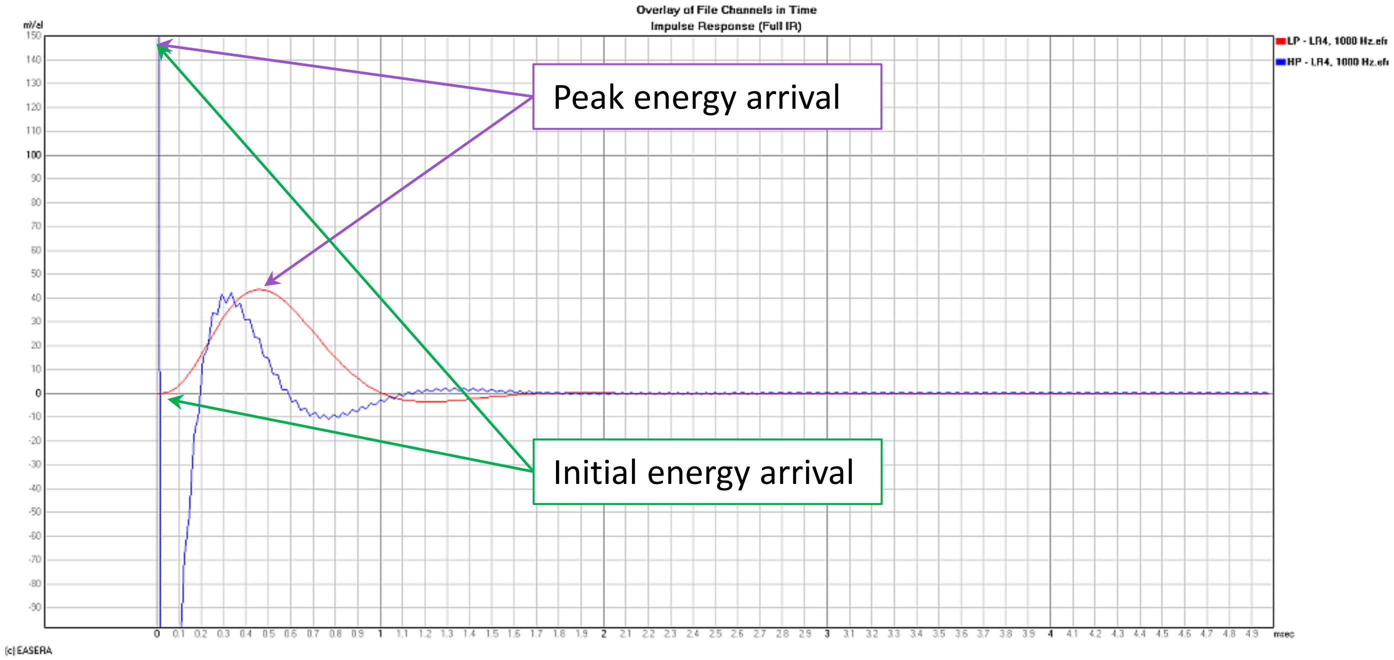
Measuring Arrival Time

Output of filters to investigate the true arrival time

Linkwitz Riley
4th order filters

High Pass
Low Pass

These filters are in temporally aligned (synchronized)

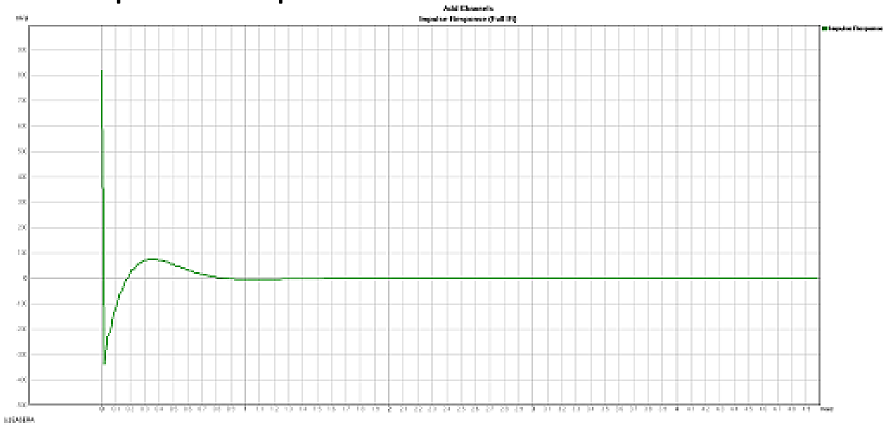


Synchronization (Alignment) of Pass Bands

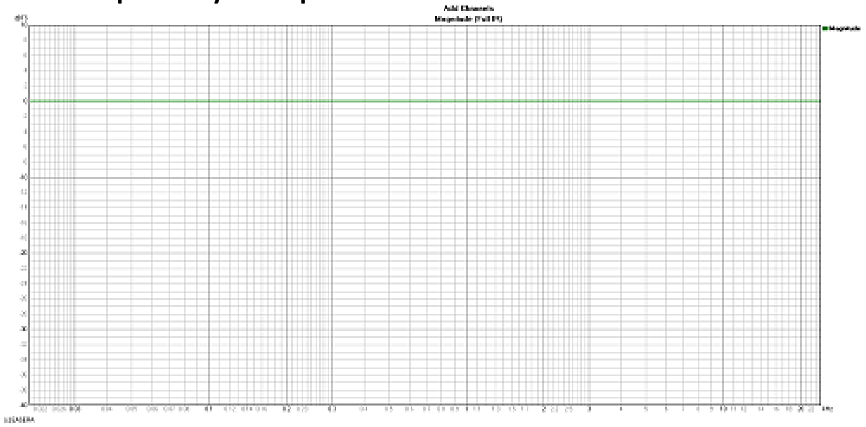
Measuring Arrival Time

Summation of the high pass and low pass filters

Impulse Response



Frequency Response



Synchronization (Alignment) of Pass Bands

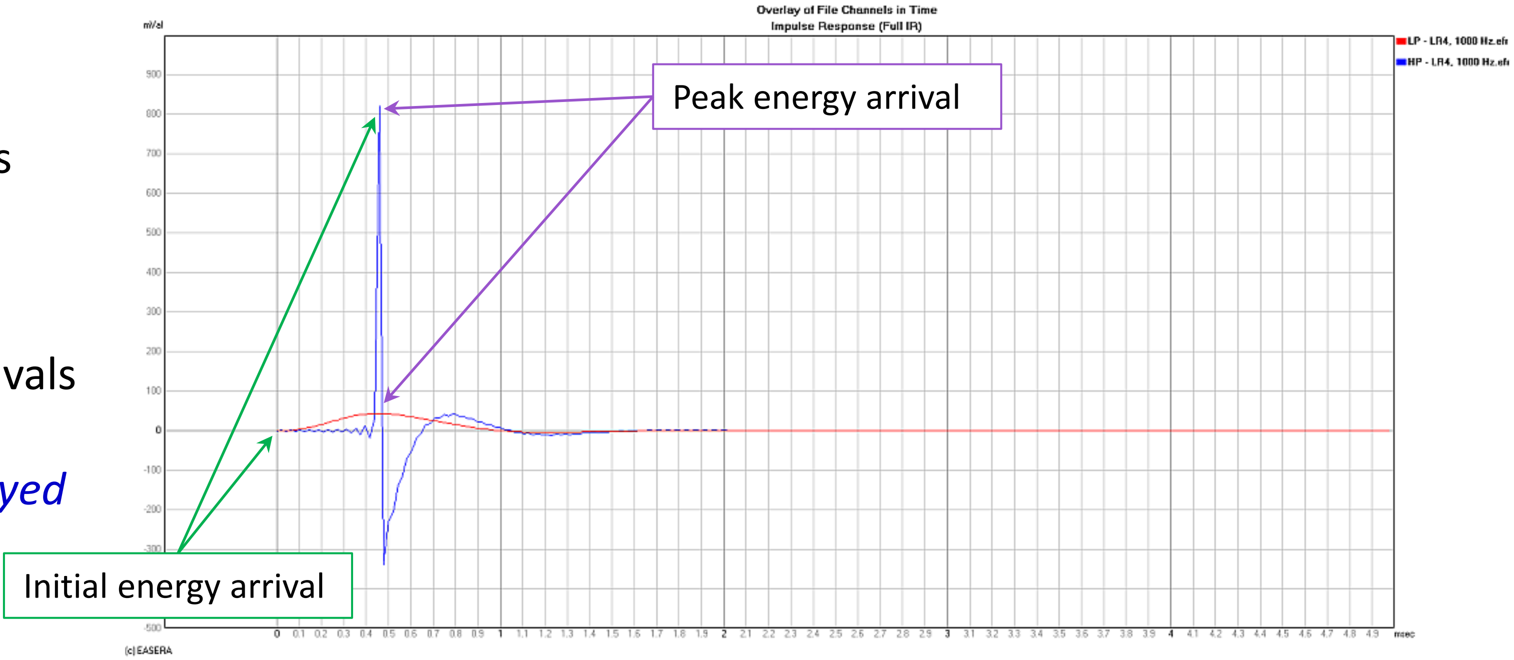
Measuring Arrival Time

Output of filters to investigate the true arrival time

Linkwitz Riley
4th order filters

High Pass
Low Pass

Peak Energy Arrivals
Not Aligned
HP filter is delayed
0.46 ms



Synchronization (Alignment) of Pass Bands

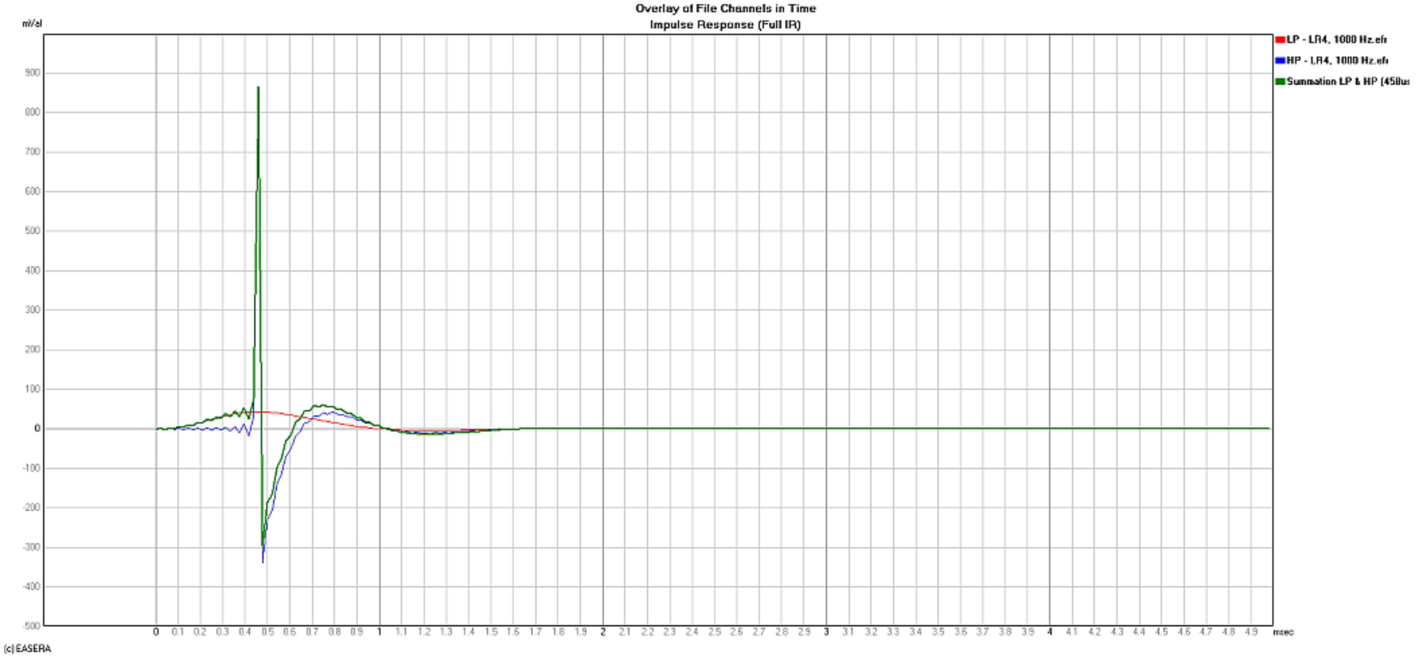
Measuring Arrival Time

Output of filters to investigate the true arrival time

Linkwitz Riley
4th order filters

High Pass
Low Pass
Summed Response

Peak Energy Arrivals
Not Aligned
HP filter is delayed
0.46 ms



Synchronization (Alignment) of Pass Bands

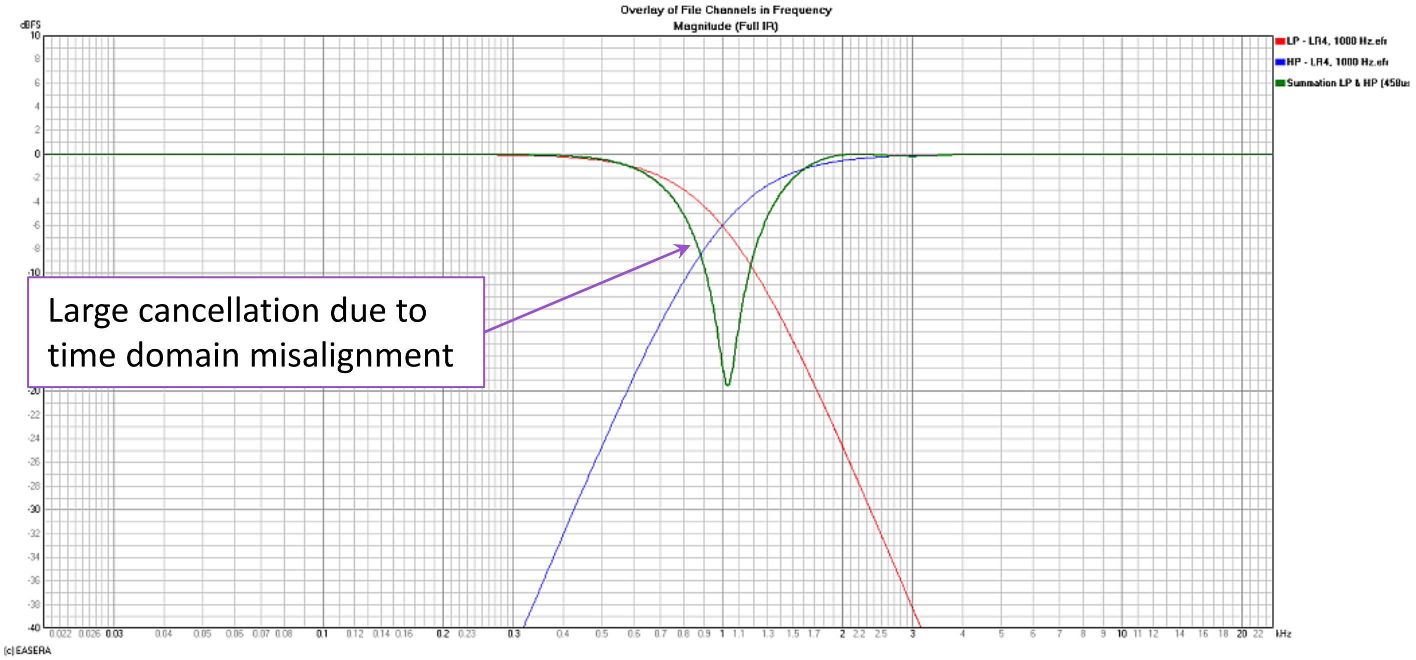
Measuring Arrival Time

Output of filters to investigate the true arrival time

Linkwitz Riley
4th order filters

High Pass
Low Pass
Summed Response

Peak Energy Arrivals
Not Aligned
HP filter is delayed
0.46 ms



Synchronization (Alignment) of Pass Bands

Measuring Arrival Time

Conclusions:

1. Phase shift is not always equivalent to delay.

2. Delay cannot be used to correct for the phase shift introduced by filters.

3. If attempting to synchronize pass bands using the matching phase method there shouldn't be any filters in the signal path during the measurements.

Any changes to the filters will change the phase response between pass bands.
May need to change the delay time to compensate for the phase change.

Synchronization (Alignment) of Pass Bands

Aligning Phase Response

Aligning the phase response of adjacent pass bands through the crossover region will yield maximum summation through the crossover region.

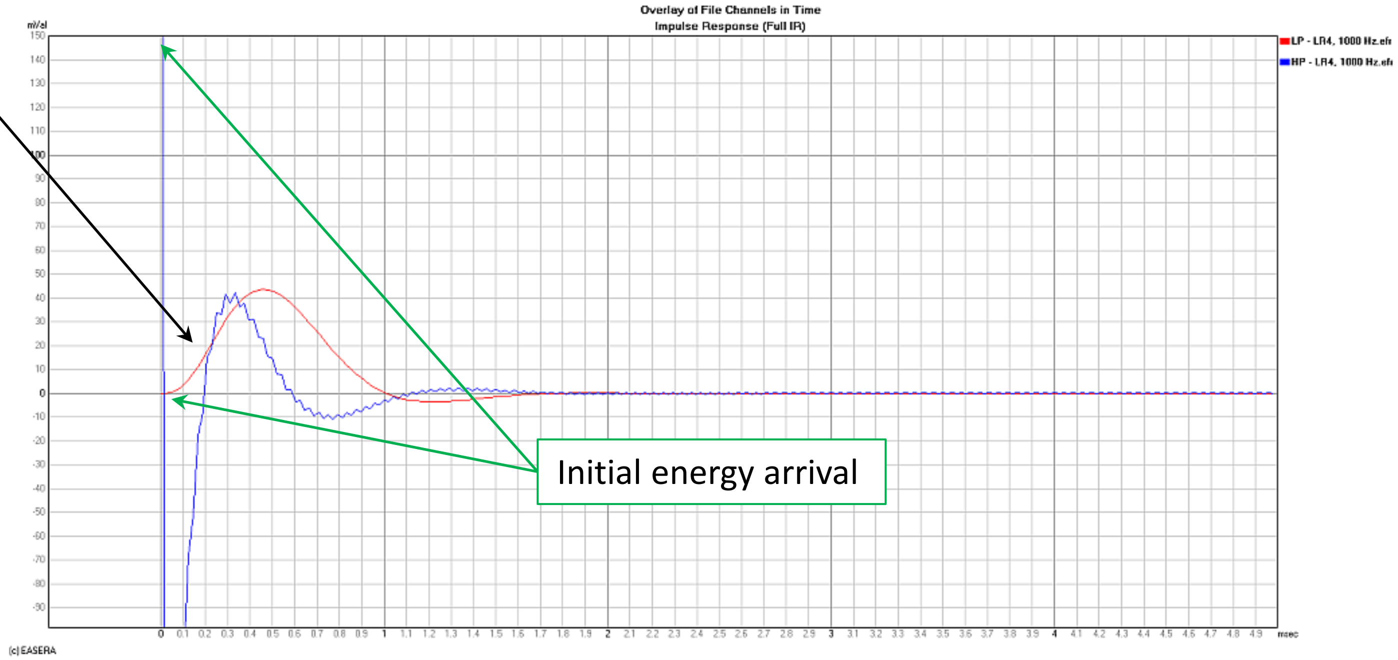
It does not guarantee time alignment of adjacent pass bands.

Synchronization (Alignment) of Pass Bands

Why Does it Look Like There is Delay?

Output of filters to investigate the true arrival time

Apparent time gap in the LP response is not due to a pure, broadband delay but rather a lack of high frequency energy content and the necessary phase shift from the low pass filter.



Synchronization (Alignment) of Pass Bands

Why Does it Look Like There is Delay?

Because the low pass filter removed high frequency information

More HF energy content in the output of a DUT increases our ability to resolve smaller time increments, $\Delta t = 1/\Delta f$

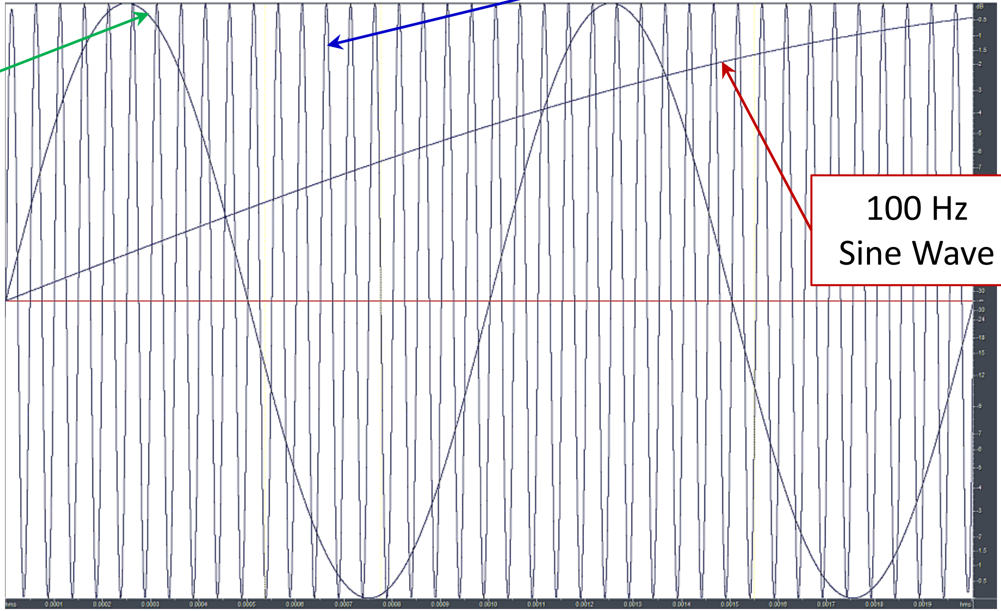
Period = 1/frequency

$P_{20kHz} = 0.05$ ms

$P_{1kHz} = 1.0$ ms

$P_{100Hz} = 10$ ms

More HF energy content in the output from a DUT ***increases the rise time of its impulse response*** at its initial arrival time.



Synchronization (Alignment) of Pass Bands

Recommended Method

1. Measure the IR of each pass band, no crossover filters.
2. Determine required delay time from IR in time domain.
3. Use these delay time values and do not change them.
4. Select the required crossover filters between adjacent pass bands to yield ***complimentary magnitude and phase*** between the adjacent pass bands.

This will result in good summation through the crossover region and synchronization of the adjacent pass bands.

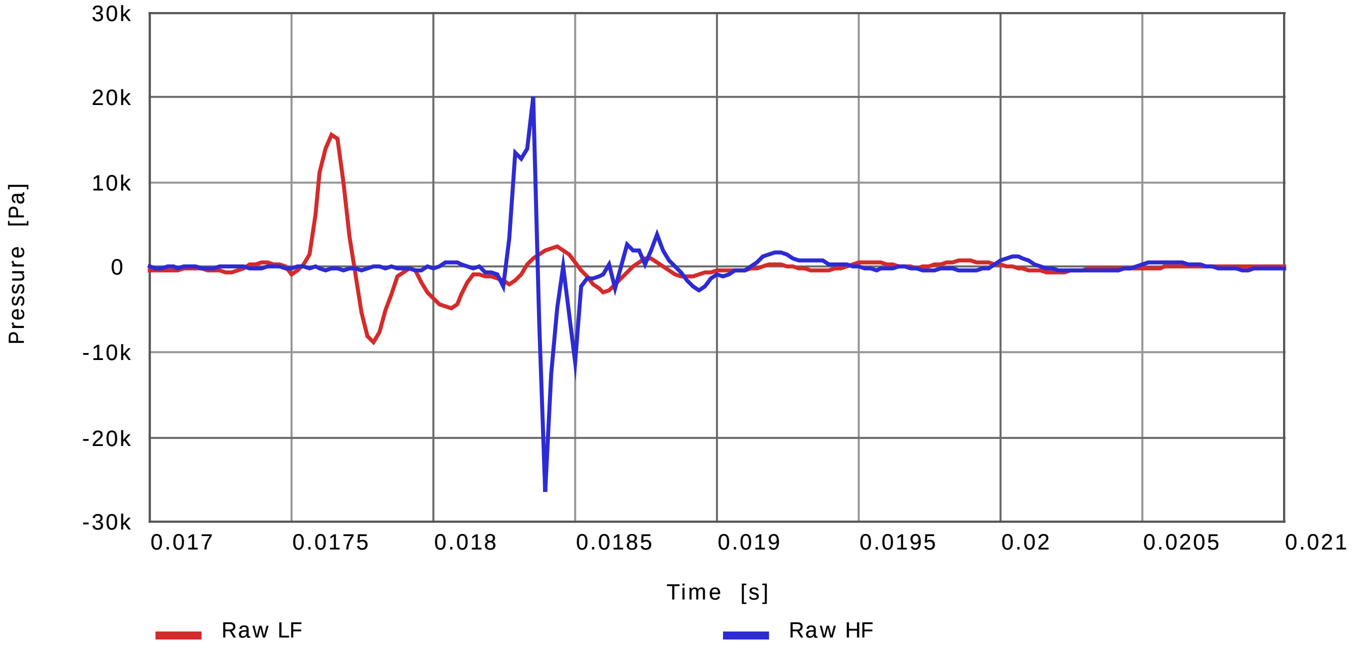
Very efficient.

Only set delay once and don't have to change if the XO filters change.

Synchronization (Alignment) of Pass Bands

Example

Impulse Response for LF and HF pass bands



Synchronization (Alignment) of Pass Bands

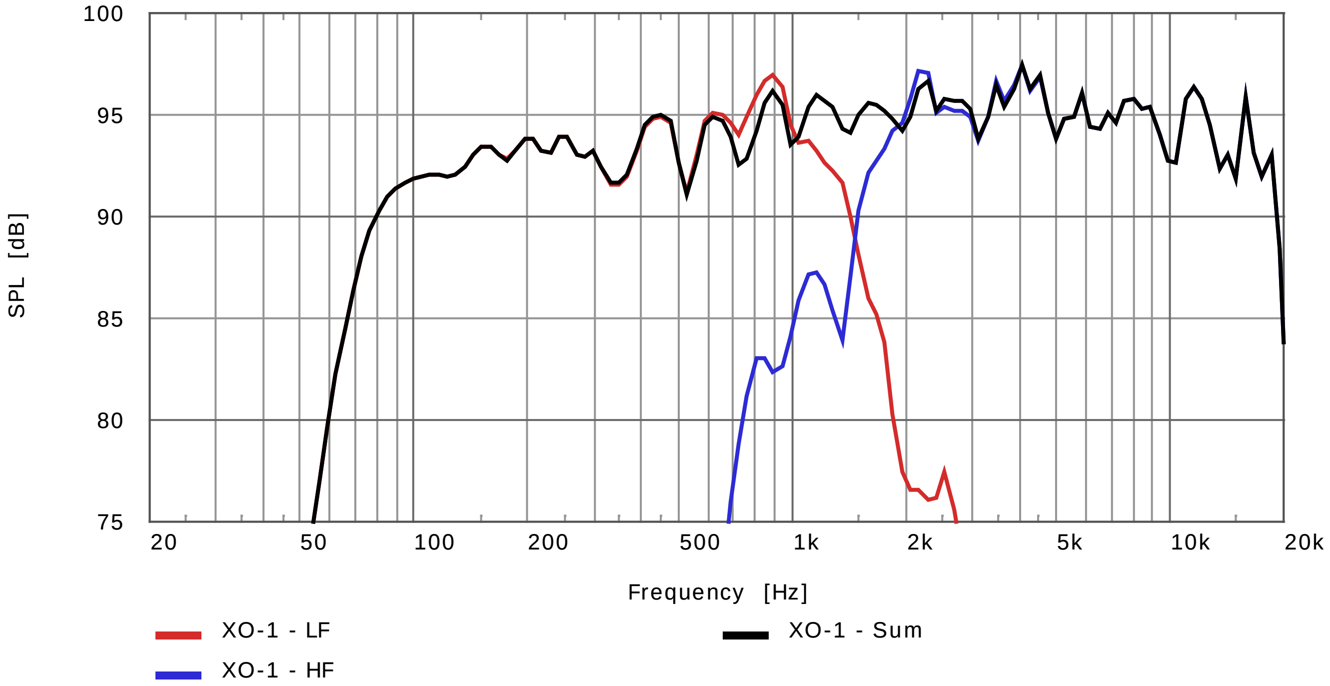
Example

High Pass: LR24 at 1.3 kHz + EQ

Low Pass: LR24 at 1.3 kHz, no delay + EQ

Original Crossover

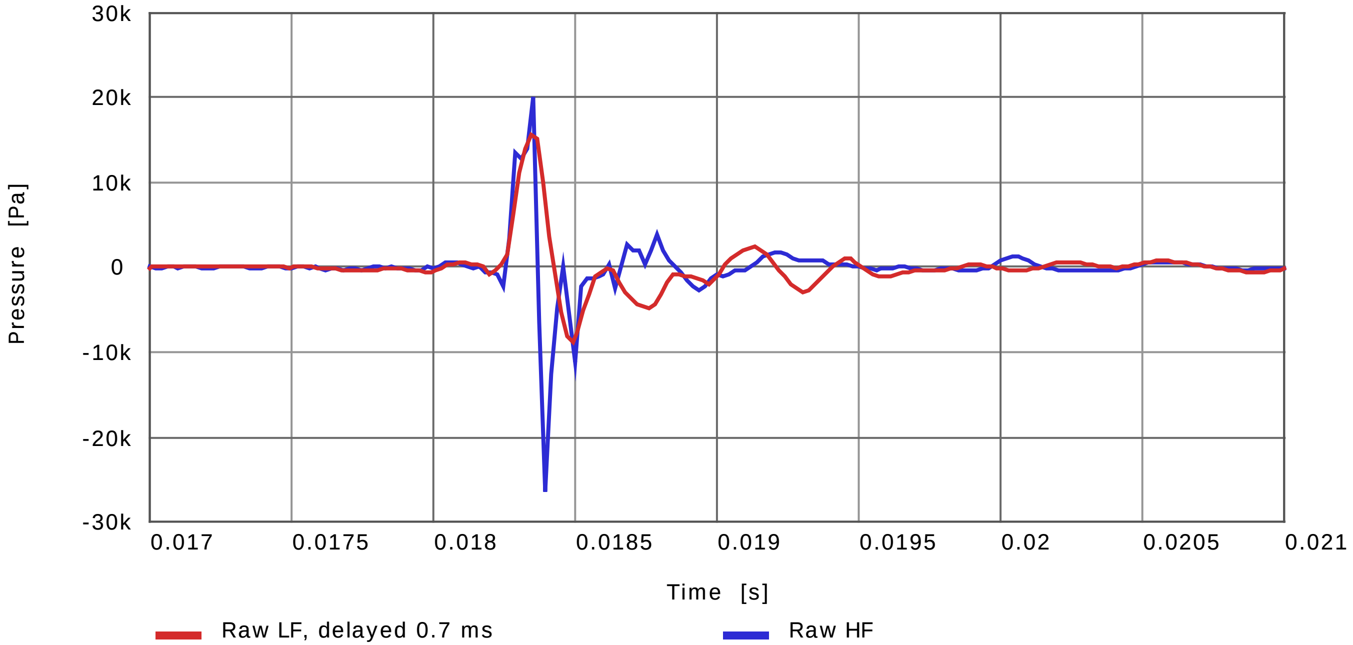
LF is not synchronized with the HF



Synchronization (Alignment) of Pass Bands

Example

LF pass band delayed 0.7 ms



Synchronization (Alignment) of Pass Bands

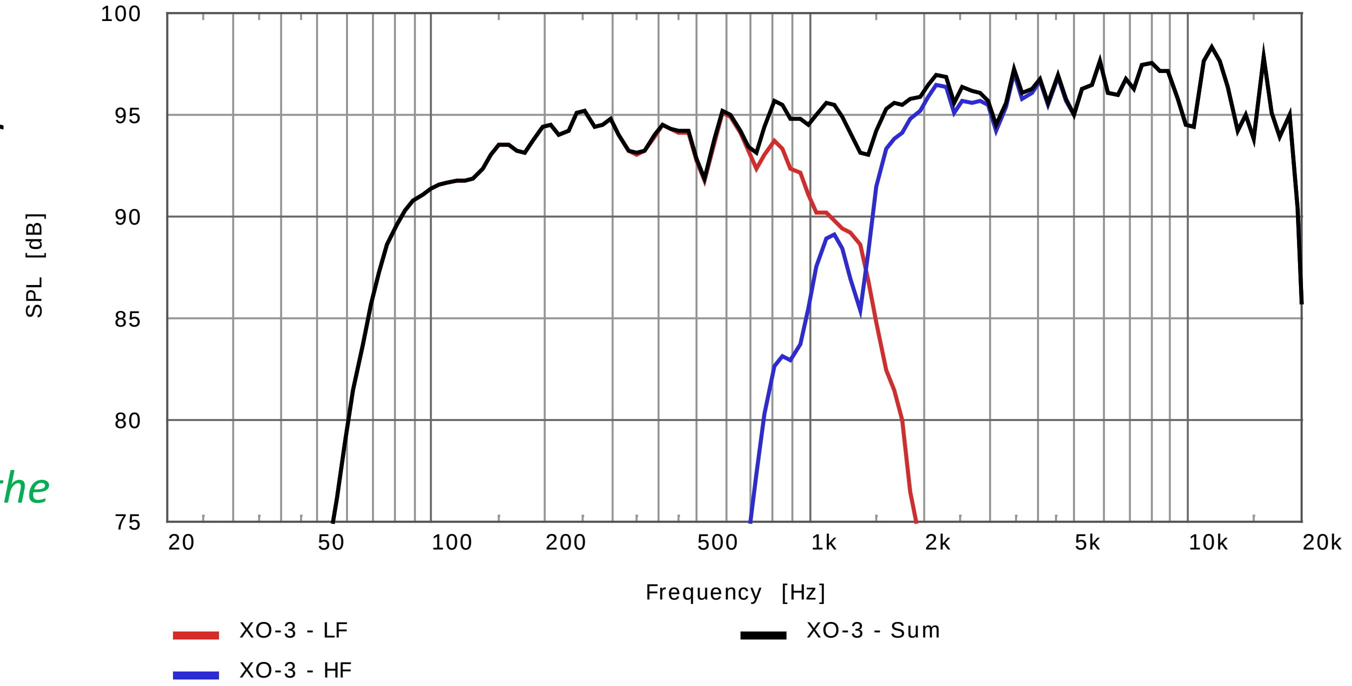
Example

High Pass: Butterworth24 at 1.1 kHz + EQ

Low Pass: Butterworth18 at 900 Hz, 0.7 ms delay + EQ

New Crossover with correct delay

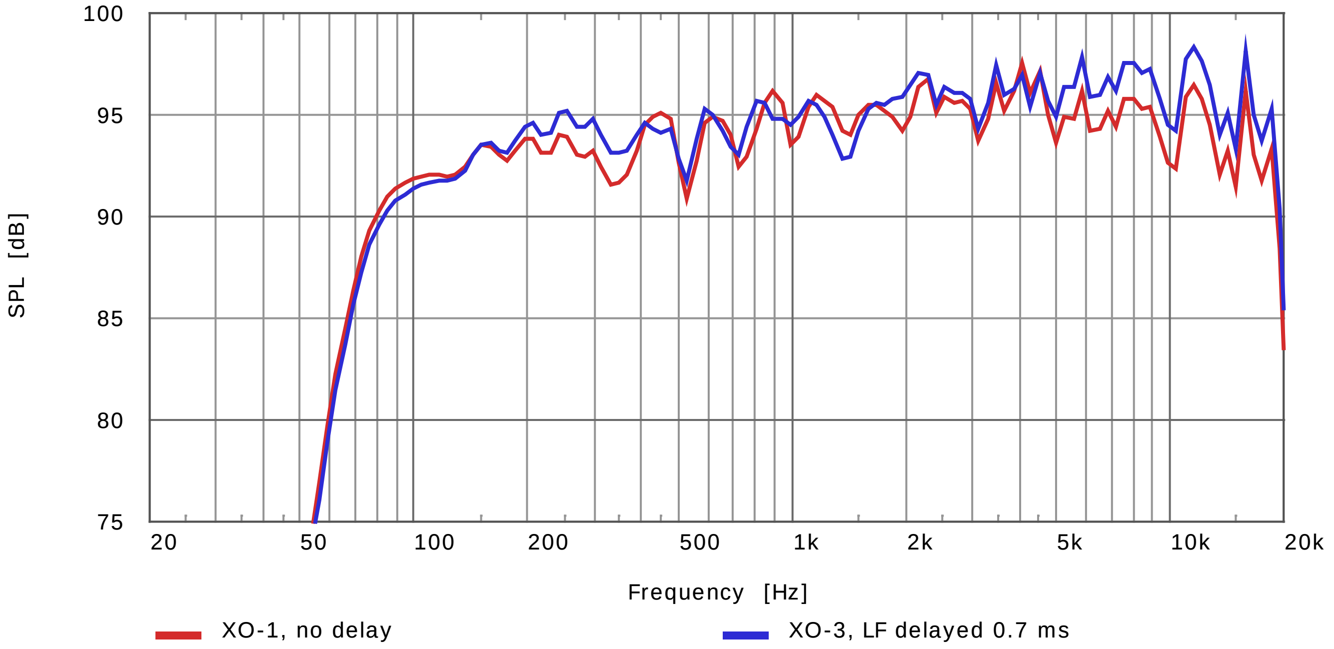
This also improved the directivity response



Synchronization (Alignment) of Pass Bands

Example

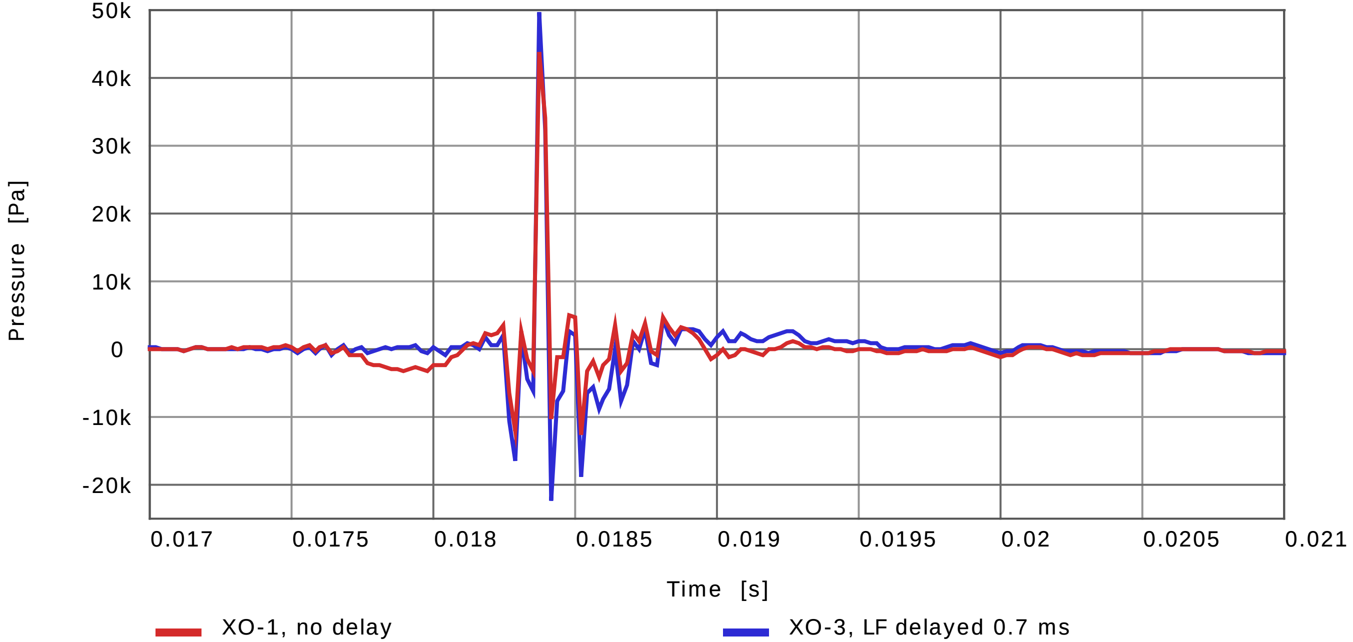
Comparison of on-axis frequency response



Synchronization (Alignment) of Pass Bands

Example

Comparison of on-axis IR



Measurement of Loudspeaker Systems

Thank you!